Avaya Aura® Communication Manager Screen Reference

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Chapter 1: Introduction

Purpose

This document provides detailed description of the fields of Communication Manager screens.

Intended audience

The primary audience for this document is:

- Avaya field technicians
- Avaya partners
- Technical support personnel
- Solution architects
- Implementation engineers
- Support personnel
- Technical support representatives

Document changes since last issue

The following change has been made to this document since the last issue:

- Added the new field Location for routing incoming overlap calls on page 147.

Related resources

Documentation

The following table lists the documents related to this product. Download the documents from the Avaya Support website at http://support.avaya.com/.

Document number	Title	Description	Audience
Design			
03-300511	*Avaya Aura® Communication Manager System Capacities Table*	Describes the system software-defined capacities information for all templates of Communication Manager, ASAI, Messaging, and Call Center.	Solution architects, Implementation engineers, Support personnel, Technical support representatives
Maintenance and Troubleshooting			
03-300431	*Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*	Describes the commands of Avaya Aura® Communication Manager that are used to monitor, test, and maintain hardware components of an Avaya server or gateway system.	Solution architects, Implementation engineers, Support personnel, Technical support representatives
Administration			
100167521	*Call Preservation Feature Description and Administration Guide*	Describes the Call Preservation feature administration for two Call Center configurations	Solution architects, Implementation engineers, Support personnel, Technical support representatives
Understanding			
555-245-205	*Avaya Aura® Communication Manager Feature Description and Implementation*	Describes the features of Avaya Aura® Communication Manager. Describes how to administer and troubleshoot features.	Solution architects, Implementation engineers, Support personnel, Technical support representatives

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Course code	Course title
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1A00234E	Avaya Aura® Fundamental Technology
AVA00383WEN	Avaya Aura® Communication Manager Overview
ATI01672VEN, AVA00832WEN, AVA00832VEN	Avaya Aura® Communication Manager Fundamentals
Docu00158	Whats New in Avaya Aura® Release 6.2 Feature Pack 2
5U00060E	Knowledge Access: ACSS - Avaya Aura® Communication Manager and CM Messaging Embedded Support (6 months)
Implementation and Upgrading	
4U00030E	Avaya Aura® Communication Manager and CM Messaging Implementation
ATC00838VEN	Avaya Media Servers and Implementation Workshop Labs
4U00115V	Avaya Aura® Communication Manager Implementation Upgrade (R5.X to 6.X)
4U00115I, 4U00115V	Avaya Aura® Communication Manager Implementation Upgrade (R5.X to 6.X)
AVA00838H00	Avaya Media Servers and Media Gateways Implementation Workshop
ATC00838VEN	Avaya Media Servers and Gateways Implementation Workshop Labs
Administration	
AVA00279WEN	Communication Manager - Configuring Basic Features
AVA00836H00	Communication Manager Basic Administration
AVA00835WEN	Avaya Communication Manager Trunk and Routing Administration
5U0041I	Avaya Aura® Communication Manager Administration
AVA00833WEN	Avaya Communication Manager - Call Permissions
AVA00834WEN	Avaya Communication Manager - System Features and Administration
5U00051E	Knowledge Access: Avaya Aura® Communication Manager Administration

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  - Scroll down Playlists, and click the name of a topic to see the available list of videos posted on the website.

Note:

Videos are not available for all products.

Support

Go to the Avaya Support website at http://support.avaya.com for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

Warranty

Avaya provides a 90-day limited warranty on Communication Manager. To understand the terms of the limited warranty, see the sales agreement or other applicable documentation. In addition, the standard warranty of Avaya and the details regarding support for Communication Manager in the warranty period is available on the Avaya Support website at http://support.avaya.com/ under **Help & Policies** > **Policies & Legal** > **Warranty & Product Lifecycle**. See also **Help & Policies** > **Policies & Legal** > **License Terms**.

Chapter 2: AAR Digit Analysis Table and ARS Digit Analysis Table

Communication Manager compares dialed numbers with the value of the **Dialed Strings** field and determines the route pattern for the dialed number.

To change the value of the fields on the AAR Digit Analysis Table screen, use the following command:

`change aar analysis` *n*, where *n* is the digit string.

To change the value of the fields on the ARS Digit Analysis Table screen, use the following command:

`change ars analysis` *n*, where *n* is the digit string.

ANI Reqd

Use this field to specify whether the R2-MFC and the Russian MF ANI calls require Automatic Number Identification (ANI).

The system displays the **ANI Reqd** field only if you disable the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related Parameters screen.

Valid entry	Usage
n	The R2-MFC or Russian MF ANI calls do not require ANI. The default value is n.
y	The R2-MFC or Russian MF ANI calls require ANI.
r	The call on the Russian Shuttle trunk or the Russian Rotary trunk drops if the ANI request fails. The system displays this option only if you enable the **Allow ANI Restriction on AAR/ARS** field on the Feature-related system parameters screen.

Related Links

Allow ANI Restriction on AAR/ARS on page 290
Request Incoming ANI (non-AAR/ARS) on page 577

Call Type (AAR only)

Use this field to enter the call type associated with each dialed string.

Valid entry	Usage
aar	Use this option for regular Automatic Alternate Routing (AAR) calls.
intl	Use this option if the Route index contains public network ISDN trunks that require international type of number encoding.
pubu	Use this option if the Route index contains public network ISDN trunks that require unknown type of number encoding.
lev0–lev2	Use this option for ISDN Private Numbering Plan (PNP) number formats.
unku	Use this option to set up an Implicit numbering plan. Users can use the plan to dial an extension without using an Automatic Route Selection (ARS) or AAR access code.

ISDN Protocol:

Call type numbering	Numbering plan identifier	Type of numbering
aar	E.164 (1)	national(2)
intl	E.164 (1)	international(1)
pubu	E.164 (1)	unknown(0)
lev0	PNP(9)	local(4)
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (ARS only)

Use this field to enter the call type associated with each dialed string.

Valid entry	Usage	China Number 1, Call Type
intl	Use this option for public-network international calls.	toll-auto
alrt	Use this option to alert attendant consoles or other digital telephones when a user places an emergency call.	normal
emer	Use this option for emergency calls.	normal
fnpa	Use this option for ten-digit North American Numbering Plan (NANP) calls.	attendant
hpna	Use this option for seven-digit NANP calls.	normal
lop	Use this option for international operator calls.	attendant
locl	Use this option for public-network local calls.	normal

Table continues…

Valid entry	Usage	China Number 1, Call Type
lpvt	Use this option local private calls.	normal
natl	Use this option for non-NANP calls.	normal
npvt	Use this option for national private calls.	normal
nsvc	Use this option for national service calls.	normal
op	Use this option for operator calls.	attendant
pubu	Use this option for public-network number (E.164)-unknown calls.	normal
svcl	Use this option for national(2) calls.	toll-auto
svct	Use this option for national(2) calls.	normal
svfl	Use this option for service call first party control calls.	toll
svft	Use this option for service call first party control calls.	local

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, x and X.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays **all**.

Related Links

Multiple Locations on page 804

Comments on this document? infodev@avaya.com

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Node Number

Use this field to specify a destination node number when the system uses node number routing or Distributed Communication System (DCS).

Valid entry	Usage
blank	Use this option if you do not want to enter the destination node number.
1 to 999	Use this option to enter the destination node number.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Route Pattern

Use this field to specify the route number that Communication Manager must use for the dialed string.

Valid entry	Usage
p1 to p2000	To enter a route index number.
digits	To enter a route pattern. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Table continues…

Valid entry	Usage
r1 to r32	To enter a Remote Home Numbering Plan Area (RHNPA) number if the dialed string requires RHNPA translations.
node	To enable the system to use node number routing.
deny	To enable the system to block a call.

Chapter 3: AAR Digit Conversion Table and ARS Digit Conversion Table

Use the AAR Digit Conversion Table and the ARS Digit Conversion Table screens to modify a dialed number for efficient routing. You can insert or delete digits from the dialed number. For example, you can delete 1 and the area code from the dialed number and avoid long distance charges by routing the call over the private network.

To change the value of the fields on the AAR Digit Conversion Table screen, use the following command:

`change aar digit-conversion` *n*, where *n* is the digit string.

To change the value of the fields on the ARS Digit Conversion Table screen, use the following command:

`change ars digit-conversion` *n*, where *n* is the digit string.

ANI Reqd

Use this field to specify whether the R2-MFC and the Russian MF ANI calls require Automatic Number Identification (ANI).

The system displays the **ANI Reqd** field only if you disable the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related Parameters screen.

Valid entry	Usage
n	The R2-MFC or Russian MF ANI calls do not require ANI. The default value is n.
y	The R2-MFC or Russian MF ANI calls require ANI.
r	The call on the Russian Shuttle trunk or the Russian Rotary trunk drops if the ANI request fails. The system displays this option only if you enable the **Allow ANI Restriction on AAR/ARS** field on the Feature-related system parameters screen.

Related Links

Allow ANI Restriction on AAR/ARS on page 290
Request Incoming ANI (non-AAR/ARS) on page 577

Conv

Use this field to enable additional digit conversion.

Del

Use this field to enter the number of digits that the system must delete from the starting of the dialed string.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays **all**.

Related Links

Multiple Locations on page 804

Matching Pattern

Use this field to specify the pattern that the system must use to match the dialed number. The pattern can be up to 18 characters long and consist of digits and the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Net

Use this field to specify the method that the call processing server must use to analyze the converted number.

Valid entry	Usage
ext	The call processing server analyzes the converted number as an extension number.
aar	The call processing server analyzes the converted number as an AAR number.
ars	The call processing server analyzes the converted number as an ARS number.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Replacement String

Use this field to specify the digit string that must replace the deleted portion of the dialed number.

Valid entry	Usage
blank	To delete the digits without replacement.
0 to 9, *	To type the digit string. The digit string can have up to 18 characters.
#	To indicate the end of dialing.

Chapter 4: Abbreviated Dialing

Use the Abbreviated Dialing screen to store and retrieve dialing lists that simplify or eliminate dialing.

The types of Abbreviated Dialing lists are:

- Enhanced list
- Group list
- Personal list
- System list

Users can use these lists to:

- Make local, long distance, and international calls.
- Activate or deactivate features.
- Access the remote system.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30
7103A Button Assignment on page 32

Enhanced List

Use this screen to create systemwide or personal lists for speed dialing.

Note:

You must activate dialing in the license file before the system programs Abbreviated Dialing Enhanced List.

To modify the value of the fields on the Enhanced List screen, use the following command:

`change abbreviated-dialing enhanced` *n*, where *n* is an enhanced list number.

Related Links

Abbreviated Dialing on page 25

DIAL CODE on page 26
Privileged on page 26
Size (multiple of 5) on page 26
Abbreviated Dialing Enhanced List on page 793

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for S8300 servers.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related Links

Enhanced List on page 25
Group List on page 27
System List on page 30

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

Group List

Use this screen to create a group list. You can assign up to 100 numbers to every group list.

To change the value of the fields on the Group List screen, use the following command:

`change abbreviated-dialing group` *n*, where *n* is the group list number.

Related Links

Abbreviated Dialing on page 25
DIAL CODE on page 26
Group List on page 28
Privileged on page 26
Program Ext on page 28
Size (multiple of 5) on page 26

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for S8300 servers.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29

System List on page 30

Group List

This field displays the number that you assign to the group list.

Related Links

Group List on page 27

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related Links

Enhanced List on page 25
Group List on page 27
System List on page 30

Program Ext

Use this field to enter the extension that has permissions to program the Group list.

Related Links

Group List on page 27

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

Personal List

Use this screen to create a personal dialing list for telephone or data module users.

To change the value of the fields on the Personal List screen, use the following command:

`change abbreviated-dialing personal` *n*, where *n* is a personal list number.

Related Links

Abbreviated Dialing on page 25
DIAL CODE on page 26
List Number on page 29
Personal List on page 30
Size (multiple of 5) on page 26

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for S8300 servers.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

List Number

This field displays which of the three personal lists is defined for the telephone.

Related Links

Personal List on page 29

Personal List

Use this field to enter the extension of the telephone using this list.

Related Links

Personal List on page 29

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

System List

Use this screen to create the system list. You can create only one system list.

To change the value of the fields on the System List screen, use the following command:

`change abbreviated-dialing system` *n*, where *n* is the system list number.

Related Links

Abbreviated Dialing on page 25
DIAL CODE on page 26
Label Language on page 31
LABELS FOR 2420/4620 STATIONS on page 32
Privileged on page 26
Size (multiple of 5) on page 26

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for S8300 servers.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

Label Language

Use this field to administer custom labels on the 2420 and 4620 telephones. The system can display the labels in one of the following languages:

- english
- french
- italian
- spanish

Note:

Unicode is not a valid language.

If you change the language, the system saves all administered labels in the new language and displays the labels in the new language.

To translate the labels to a different language, type `user-defined` in the **Label Language** field. You can use Avaya Message Editing Tool (AMET) to assign translations to the labels. For more information about AMET, see *Job Aid: Telephone Message Translations*.

Related Links

System List on page 30
Display Character Set on page 787

LABELS FOR 2420/4620 STATIONS

Use this field to customize the labels for the Abbreviated Dialing buttons on the 2420 and 4620 telephones. The labels can have up to 15 alphanumeric characters.

Related Links

System List on page 30

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related Links

Enhanced List on page 25
Group List on page 27
System List on page 30

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related Links

Enhanced List on page 25
Group List on page 27
Personal List on page 29
System List on page 30

7103A Button Assignment

Use this screen to assign abbreviated dialing numbers to the 7103A telephone buttons. You can use this list only for 7103A fixed feature telephones. You can implement only one 7103A abbreviated dialing list and this list applies to all 7103A fixed feature telephones in the system.

To modify the value of the fields on the 7103A Button Assignment screen, use the following command:

`change abbreviated-dialing 7103A-buttons` *n* where *n* is a 7103A button assignment abbreviated dialing list.

Related Links

Abbreviated Dialing on page 25
DIAL CODE on page 33

DIAL CODE

Use this field to assign a number to each dial code button. Any additions or changes apply to all 7103A fixed feature telephones. A call progress tone receiver is tied up while the system is waiting. The outgoing calling capability is impaired as there are a limited number of receivers in the system.

You can also assign a VDN.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for S8300 servers.

Related Links

7103A Button Assignment on page 32

Chapter 5: Access Endpoint

Use the Access Endpoint screen to administer an access endpoint or a Wideband Access Endpoint (WAE).

WAE is an endpoint application connected to line-side non-ISDN T1 or E1 facilities. WAEs do not have signaling interface with the system. You can administer WAE only if **Wideband Switching** is enabled for the system.

An access endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. You can use access endpoints to assign an extension number to a single trunk and eliminate the need to dedicate an entire trunk group for a single trunk. You can specify an access endpoint as the originator or the destination endpoint of an administered connection.

Note:

Access endpoints and WAEs use the same resources that the trunks use. Therefore, the sum of access endpoints and trunks cannot exceed the maximum number of trunks available in the system.

To change the value of the fields on the Access Endpoint screen, use the following command:

`add access-endpoint` *next*, where *next* is the next available extension.

Related Links

Wideband Switching on page 807

Communication Type

Use this field to specify the communication type for the access endpoint.

Valid entry	Usage
56k-data	For a DS1 access endpoint. You can use this option for the robbed-bit trunks. The default value is 56k-data.
64K-data	For a DS1 access endpoint. Do not use this option for the robbed-bit trunks.
voice-grade-data	For an analog tie trunk access endpoint.
wideband	For a WAE.

COR

Use this field to assign a class of restriction (COR) number to the access endpoint. You must administer this field so that only an Administered Connection (AC) endpoint can be connected to another AC endpoint.

COS

Use this field to assign a class of service (COS) number to the access endpoint. You must administer this field so that Communication Manager restricts the use of the **Call Forwarding All Calls** feature on the access endpoint.

Extension

Use this field to assign an extension number to the nonsignaling trunk.

ITC

Use this field to assign a transmission facility to the access endpoint. The system displays this field only if the **Communication Type** field is set to 56k-data, 64k-data, or Wideband.

Valid entry	Usage
restricted	Restricted transmission facilities are used to complete the call. Before using this option, ensure that the associated port of the access endpoint is a channel of a DS1 circuit pack and the **Line Coding** field of the DS1 circuit pack is set to `B8ZS`. The default value is restricted.
unrestricted	Unrestricted transmission facilities are used to complete the call.

Name

Use this field to provide a name for the access endpoint.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

(Starting) Port

Use this field to assign a port number to the access endpoint.

You can assign the port number by using one the following formats:

- *cabinet number* : *carrier number* : *slot number* : *circuit number*

Valid entry	Usage
01 to 64	The first and second characters represent the cabinet number.
A to E	The third character represents the carrier number.
01 to 20	The fourth and fifth characters represent the slot number.
01 to 04 (Analog TIE trunks)	The sixth and seventh characters represent the circuit number.
01 to 31	

- *gateway number* : *module number* : *circuit number*

Valid entry	Usage
001 to 250	The first three characters represent the gateway number.
V1 to V9	The fourth and fifth characters represent the module number.
01 to 31	The sixth and seventh characters represent the circuit number.

Note:

For WAEs, use the DS1 circuit packs, not analog tie trunks.

Related Links

Signaling Mode on page 223

TN

Use this field to assign a tenant number to the access endpoint. You can enter a value from 1 to 250.

Width

The system displays this field only if the **Communication Type** field is set to wideband. Use this field to provide the number of adjacent DS0 ports beginning with the specified starting port that make the WAE. You can enter a value between 2 to 31. The default entry is 6.

Related Links

Communication Type on page 34

Chapter 6: Administered Connection

Use the Administered Connection screen to assign an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The system establishes the AC after a restart or when the AC is active.

To change the value of the fields on the Administered Connection screen, use the following command:

`change administered-connection` *n*, where *n* is the AC number.

Connection Number

This field displays the AC number.

Destination

Use this field to specify the address of the destination access endpoint or data endpoint. The endpoint can be remotely connected to the server on which the AC is assigned.

You can enter one of the following:

- A local extension that is assigned to an access endpoint or a data endpoint
- An Abbreviated Dialing entry

The value must be consistent with the dial plan.

Enable

Use this field to enable Communication Manager to establish the AC when the AC is active.

Valid entry	Usage
y	Communication Manager attempts to establish the AC when the AC is active. The default value is y.
n	Communication Manager does not attempt to establish the AC when the AC is active.

Name

Use this field to enter the name of the AC. You can enter up to 27 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Originator

Use this field to specify an assigned access endpoint extension or data module extension. The endpoint must be registered with the same instance of Communication Manager on which you administered the nonsignaling DS1 trunk or the analog tie trunk.

AUTHORIZED TIME OF DAY

Continuous

Use this field to schedule the AC to be active for the entire time. The AC is reestablished if the connection fails.

Valid entry	Usage
y	The AC is active for the entire time. The default value is y.
n	The AC is active only for the scheduled days and times.

Duration

Use this field to specify the duration for which the scheduled AC must remain active. The maximum duration is 167:59. The system displays this field only if you set the **Continuous** field to n.

Valid entry	Usage
000 to 167	To enter the duration in hours.
00 to 59	To enter the duration in minutes.

Start Days (Sun through Sat)

Use the following fields to specify the days when an attempt is made to establish the AC:

- **Sun**
- **Mon**
- **Tue**
- **Wed**
- **Thu**
- **Fri**
- **Sat**

The system displays these fields only if you set the **Continuous** field to n. The start days can be the days when the AC is inactive. A scheduled AC might be active over a number of days. In this situation, use the fields only to specify the days when the AC starts.

Related Links

Continuous on page 39

Start Time

Use this field to specify the time of day when Communication Manager must establish the AC. You can enter a value from 00:00 to 23:59. The system displays this field only if you set the **Continuous** field to n.

Related Links

Continuous on page 39

MISCELLANEOUS PARAMETERS

Alarm Threshold

Use this field to specify the number of attempts to establish or reestablish an AC that must fail consecutively before the system generates an AC alarm. For example, If you set this field to 1, the system generates an alarm after the first failure. You can enter a value from 1 to 10. The default value is 5.

The system displays this field only if you set the **Alarm Type** field to a value other than none.

Related Links

Alarm Type on page 41

Alarm Type

Use this field to specify the type of alarm generated when:

- The AC cannot be established.
- The AC cannot be reestablished after a failure.
- The number of consecutive failures equals the alarm threshold.

Valid entry	Usage
warning	For failures that do not cause significant degradation of service. These alarms are not reported to the attendant console or Initialization and Administration System (INADS).
major	For failures that cause critical degradation of service and require immediate attention.
minor	For failures that: • Cause degradation of service, but continue to support a crucial part of the system. • Require action, but the consequences of the failure are gradual. • Interfere with a feature or impair the services to trunks or stations.
none	For disabling the alarm notification.

Auto Restoration

Use this field to enable the system to reestablish the AC that failed. The **Auto restoration** field is available only for the AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group.

Priority

Use this field to specify the order in which the ACs are established. You must enter a value from 1 to 8. The highest priority is 1, and the lowest priority is 8. The default priority is 5.

Retry Interval

Use this field to specify the duration between the attempts to establish or reestablish the AC. You must enter the duration in minutes. The default value is 2 minutes.

Chapter 7: Agent Login ID

In an Expert Agent Selection (EAS) environment, use the Agent Login ID screen to:

- Add an agent login ID.
- Change the options of an agent login ID.
- Assign skills to an agent login ID.

To change the value of fields on the Agent Login ID screen, use one of the following commands:

- **change agent-loginID** *n*, where *n* is the agent login ID.

  For any changes to take effect, the agent must log out and log in again.

- **change agent-loginID** *n* auto, where *n* is the agent login ID.

  For the changes related to the skill set to take effect, the system logs out and logs in the agent. However, for the other changes to take effect, the agent must log out and log in again.

Agent Login ID: page 1

AAS

Use this field to enable the system to use the agent login ID as a port for Auto Available Split/Skill (AAS).

Valid entry	Usage
n	To use the agent login ID as a port for the switch adjunct equipment.
y	To use the agent login ID as a port for AAS. If you use this option, you must run the **remove Agent-LoginID** command.

ACW Agent Considered Idle

Use this field to include the After Call Work (ACW) agents in the Most-Idle Agent (MIA) queue.

Valid entry	Usage
system	To apply the systemwide values.
n	To exclude the ACW agents from the MIA queue.
y	To include the ACW agents in the MIA queue.

Related Links

ACW Agents Considered Idle on page 318

AUDIX

Use this field to enable the system to use the agent login ID as a port for voice messaging.

If an agent login ID is used as a port for AAS, the agent login ID cannot be used as a port for voice messaging.

Audix Name for Messaging

Use this field to assign a name to the messaging system that is used for Leave Word Calling (LWC) reception. The messaging system also provides call coverage to the agent login ID.

Auto Answer

Use this field to enable the **auto answer** setting of the agent login ID.

Valid entry	Usage
station	The **auto answer** setting on the station overrides the **auto answer** setting of the agent.
acd	The **auto answer** setting applies only to the ACD and direct agent calls. The agent extension gets a ringing tone for non-ACD calls.
none	The **auto answer** setting is disabled. The agent extension gets a ringing tone for all calls.
all	The **auto answer** setting applies to all ACD and non-ACD calls to the agent. For non-ACD calls, the agent extension receives a single ring. To prevent the ring, use the **ringer-off** button.

Related Links

Allow Ringer-off with Auto-Answer on page 324
Auto Answer on page 704

Aux Work Reason Code Type

Use this field to specify whether the agent must enter a reason code when changing to the AUX work mode.

Valid entry	Usage
system	The systemwide settings apply.
none	The agent must not enter a reason code when changing to the AUX work mode.
requested	The agent can enter a reason code when changing to the AUX work mode.
forced	The agent must enter a reason code when changing to the AUX work mode.

Related Links

Aux Work Reason Code Type on page 327
Reason Codes on page 809

COR

Use this field to assign a class of restriction (COR) number to the agent login ID. You can enter a value from 0 to 995. The default value is 1.

Coverage Path

Use this field to specify the coverage path used by the calls to the agent.

The coverage path is used when:

- The agent is logged out.
- The agent does not answer.
- The agent is attending a personal call while logged in.

Valid entry	Usage
blank	Use this option if you do not want to enter a coverage path.
1 to 199	Use this option to enter a coverage path number. The upper limit might change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.
t1 to t199	Use this option to enter a time-of-day table number.

Direct Agents Calls First

Use this field to enable the system to select the direct agent calls before other ACD calls. The system displays this field only if you set the **Call Handling Preference** field to `percent-allocation`.

Related Links

Call Handling Preference on page 50

Forced Agent Logout Time

Use this field to specify the time at which the system must log out the agent.

Valid entry	Usage
blank	Use this option if you do not want to specify the time at which the system must log out the agent.
00, 15, 30, or 45	Use this option to enter the time in minutes.
01 to 23	Use this option to enter the time in hours.

Related Links

Timezone Offset on page 541

Login ID

This field displays the login id for the agent.

LoginID for ISDN Display

Valid Entry	Usage
y	The Agent LoginID CPN (Calling Party Number) and **Name** are included in ISDN messaging over network facilities.
n	Default. The calling party name and number are determined by the conditions administered for the trunk group.

Related Links

Name on page 48

Logout Reason Code Type

Use this field to specify whether the agent must enter a reason code when logging out.

Valid entry	Usage
system	The systemwide settings apply.
none	The agent must not enter a reason code when logging out.
requested	The agent can enter a reason code when logging out.
forced	The agent must enter a reason code when logging out.

Related Links

Logout Reason Code Type on page 327
Reason Codes on page 809

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related Links

AUDIX Name on page 389

Maximum time agent in ACW before logout (sec)

Use this option to enter the maximum time an agent can be in ACW on a per agent basis.

Valid entry	Usage
system	With this option, system-wide settings apply. This is the default option.
none	With this option, ACW timeout is inapplicable to the agent.
30 to 9999	With this option, you can provide the number of seconds in a specific timeout period. The setting takes precedence over the system setting for maximum time in ACW.

Related Links

Maximum Time Agent in ACW before Logout (sec.) on page 330

Messaging Server Name for Messaging

Use this field to assign a name for the messaging system that is used for LWC reception and that provides coverage for an agent login id. By default, the field is blank.

MIA Across Skills

Use this field to specify if the system must remove an agent from the MIA queue.

Valid entry	Usage
system	With this option, system-wide settings apply. This is the default option.
y	With this option, the system removes an agent from the MIA queues for all the splits or skills for which an agent is available when the agent answers a call from any assigned splits or skills.
n	With this option, the system removes ACW agents from the queue.

Related Links

MIA Across Splits or Skills on page 320

Name

Use this field to enter the name of the agent. You can enter up to 27 alphanumeric characters.

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Display Character Set on page 787

Password

Use this field to assign a password to the agent login ID.

The **Minimum Agent-LoginID Password Length** field specifies a minimum length for the password.

Related Links

Minimum Agent-LoginID Password Length on page 314

Password (enter again)

Use this field to enter the same password that you entered for the **Password** field.

Port Extension

Use this field to assign an extension to the port used for AAS or voice messaging.

You cannot use a vector directory number (VDN) or an agent login ID as a port extension.

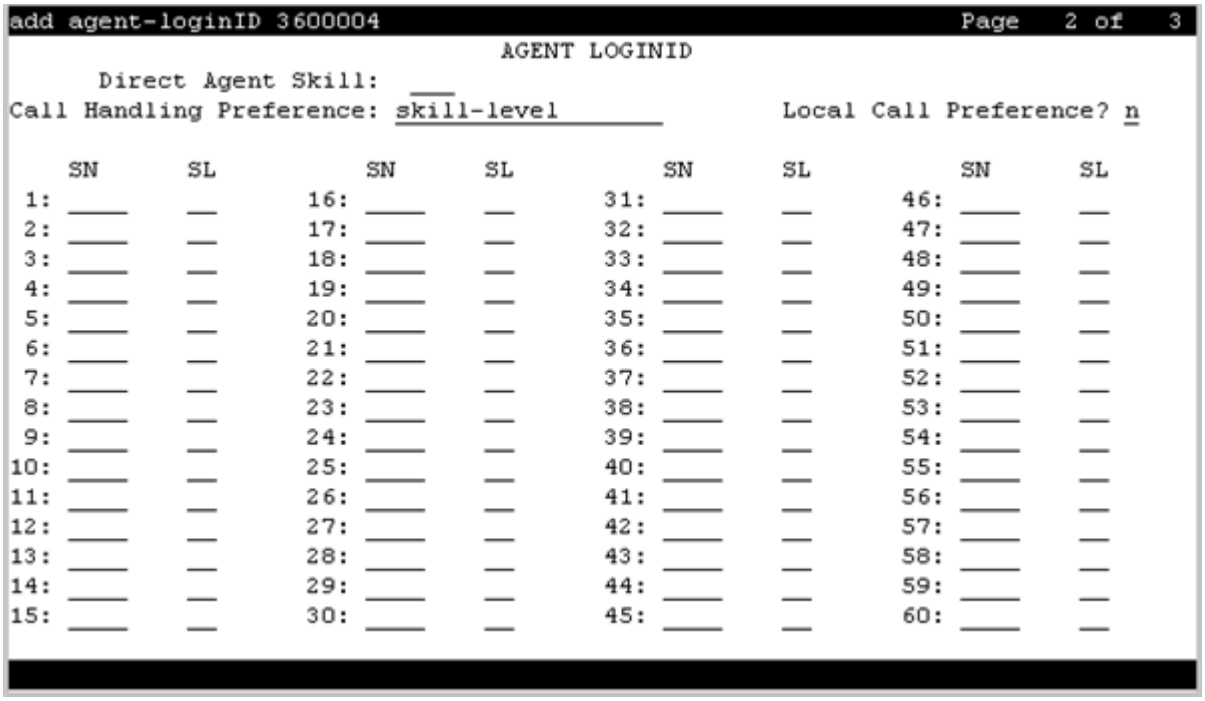
Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

TN

Use this field to assign a tenant number (TN) to the agent login ID. The default value is 1. The upper limit might change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Agent Login ID: page 2



Call Handling Preference

Use this field to specify the call that the agent must receive when calls are in queue.

Valid entry	Usage
skill-level	For the oldest high priority call waiting for an agent with the highest skill level.
greatest-need	For the oldest high priority call waiting for an agent with any skill level.
percent-allocation	For the call waiting for an agent with a skill that deviates the most from the administered allocation. This option is available only with Avaya Business Advocate.

Direct Agent Skill

Use this field to specify the skill number used to handle direct agent calls. You can enter a value from 1 to 99. The upper limit might change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Local Call Preference

Use this field to enable the Local Call Preference feature. The system displays this field only if you set the **Multiple Locations** field to y. For calls queued in more than one skill for a multiskilled EAS agent, the system gives preference to the calls whose trunk location number matches the location number of the previously busy agent. This feature is available only with Call Center Release 3.0 and later.

Related Links

Call Center Release on page 808
Multiple Locations on page 804

PA

Use this field to allocate a percentage of the agent time for each skill that you assign to the agent. The total allocation for all agent skills must be equal to 100%.

RL

Use this field to administer the agent as a reserve agent for a skill or a group of skills. You can assign a reserve level or an interruptible level.

The system displays this field only if you enable the **Business Advocate** feature.

Valid entry	Usage
1 or 2	Use this option to assign a reserve level to the agent.
a	Use this option to assign the auto-in-interrupt interruptible level to the agent.
m	Use this option to assign the manual-in-interrupt interruptible level to the agent.
n	Use this option to assign the notify-interrupt interruptible level to the agent.
blank	Use this option if you do not want to assign a reserve level or an interruptible level to the agent.

Related Links

Interruptible Aux Threshold on page 387

Service Objective

Use this field to enable the Service Objective feature. The agent receives calls according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and administration of the Service Objective feature.

The Service Objective feature is available only with Avaya Business Advocate. The system displays this field only when you set the **Call Handling Preference** field to `greatest-need` or `skill-level`.

Related Links

Call Handling Preference on page 50

SL

Use this field to specify a skill level for every skill that you assign to the agent.

If you disable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to two priority levels. If you enable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to 16 priority levels.

SN

Use this field to assign a skill number to the agent.

If you disable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to four skills. If you enable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to 20 or 60 skills depending on the system configuration.

Chapter 8: Alias Station

Use the Alias Station screen to:

- Administer new telephone types that are not supported by the system.
- Map new telephone types to a telephone type that supports Alias Station.
- Assign names to telephone types.

To change the value of the fields on the Alias Station screen, use the following command:

```
change alias station
```

Alias Set Type

Use this field to assign a name to the nonsupported telephone type. You can enter up to five alphanumeric characters.

Supported Set Type

Use this field to enter a supported telephone type. For example, DCP, H.323, SIP, or analog telephone types.

Chapter 9: Alphanumeric Dialing Table

Use the Alphanumeric Dialing Table screen to associate Alpha-name to digit strings. Users can make a data call by entering the Alpha-name instead of the digit string.

To change the value of the fields on the Alphanumeric Dialing Table screen, use the following command:

`change alphanumeric-dial-table`

Alpha-name

Use this field to assign a unique name to the digit string. The name must begin with an alphabet and can have up to eight alphanumeric characters.

Mapped String

Use this field to enter a string that you must use to generate the dialing string. You can enter up to 24 characters. You can also use the facility access codes in the string.

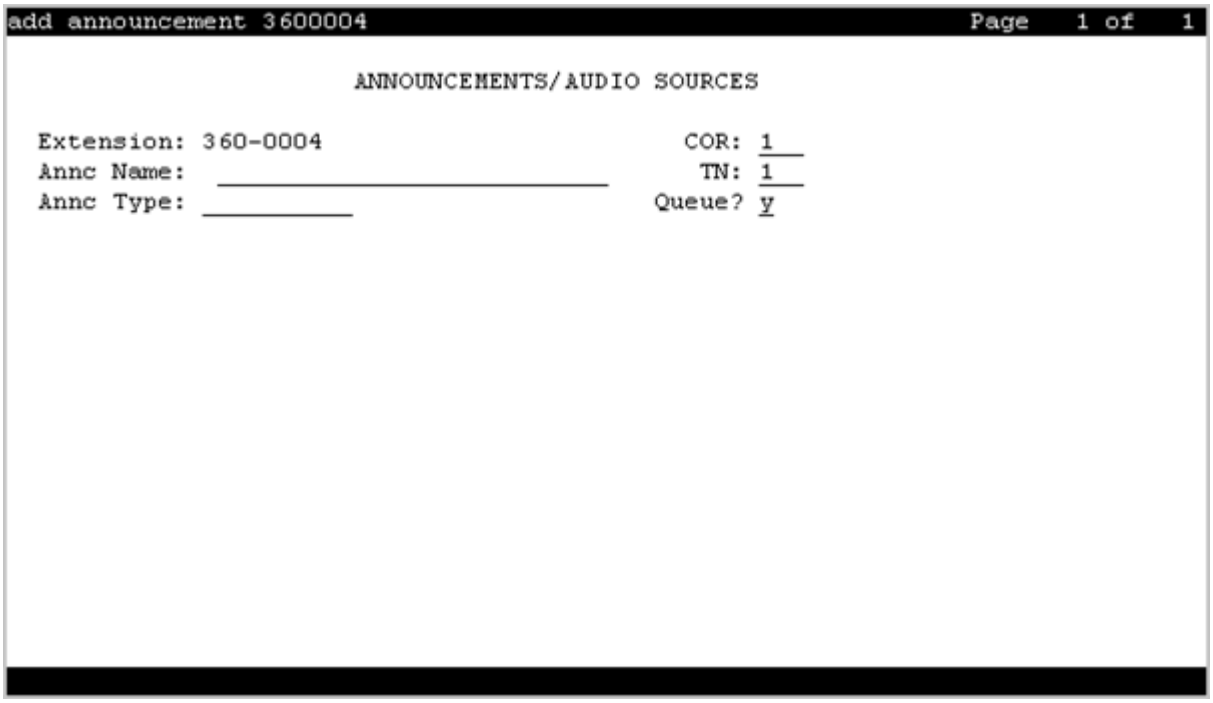
Valid entry	Usage
0 to 9	To enter a digit.
A to Z, a to z	To enter an alphabet. The system maps the uppercase alphabets to the lowercase alphabets.
(,), /, -, ^, space	To enter a special character.
+	To enable the system to wait for a dial tone.
%	To indicate that rest of the digits are for end-to-end signaling.
,	To enable the system to pause for 1.5 seconds.
#	To enter a DTMF digit pound.
*	To enter a DTMF digit asterisk.

Chapter 10: Announcements/Audio Sources

Use this screen to assign announcements to circuit packs and port locations.

To change the value of the fields on the Announcements/Audio Sources screen, use the following command:

`add announcement` *n*, where *n* is the extension number.



Annc Name

Use this field to assign a name to the announcement. You can enter up to 27 characters. Announcement names on a single VAL circuit pack must be unique. You can enter alphabets, digits, and the following special characters: /, :, *, ?, <, >, \\, and space.

The system saves the value of this field as the filename of the announcement.

Comments on this document? infodev@avaya.com

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Annc Type

Use this field to specify the type of announcement.

Valid entry	Usage
analog	To play the announcement from an external device for a specific period and then disconnect. The caller hears a click when the device is disconnected.
analog-m	To play the announcement continuously from an external announcement device.
analog-fd	To play the announcement from an external device for a specific period and then disconnect. The caller hears a click when the device is disconnected.
aux-trunk	To play the announcement from an external announcement device with a 4-wire aux interface.
aux-trk-m	To play the announcement from an external announcement device that does not indicate that the playback is active.
ds1-fd	To play the announcement from a DS1 port on the circuit pack. The caller does not hear a click when the device is disconnected. The system sends a disconnect message to the port to stop the playback when the announcement is complete.
ds1-ops	To play the announcement from a DS1 port on the circuit pack. The caller does not hear when the device is disconnected.
ds1-sa	To play the announcement from a DS1 port on the circuit pack. The caller does not hear a click when the device is disconnected. The system sends a disconnect message to the port to stop the playback when the announcement is complete.
integrated	To play the announcement stored internally on a special integrated announcement circuit pack. Use this option for general and VDN announcements.
integ-mus	To play the announcement from an integrated music source.
integ-rep	To play the announcement with integrated repeating.

COR

Use this field to assign a class of restriction (COR) number to the announcement. You can enter a value from 0 to 995.

Extension

Use this field to assign an extension number to the announcement.

Note:

If you enter a shortened extension from the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Group/Port

Use this field to assign a board location or an audio group number to the announcement.

For an integrated announcement, the system displays the field name as **Group/Board**.

You must enter the value by using one of the following formats:

- G*nn*, where *nn* is an audio group number.
- *aaxss*, where *aa* is a cabinet number, *x* is a carrier number, and *ss* is a slot number.
- *ggg*v9, where *ggg* is a gateway number.

Note:

To administer DID Intercept announcements in a multilocation system where each location requires a different announcement, you must enter an audio group number instead of a VAL port number.

Protected

Use this field to set the protection mode for an integrated announcement.

Valid entry	Usage
y	A user cannot change or delete the integrated announcement using a telephone session or an FTP session. When you record or transfer an announcement file by using FTP, the system sets the announcement file to read only.
n	A user can change or delete the integrated announcement using a telephone session or an FTP session.

Comments on this document? infodev@avaya.com

Queue

Valid Entry	Usage
y	Queues calls for the announcement if the announcement type is integrated, integ-rep or aux-trunk. The caller is always connected to the beginning of the announcement. ACD, vectoring delay announcements and call centers should always use this option. This is the default.
n	No queue nor barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is unavailable.
b	Sets up barge-in if the announcement type is integrated, integ-rep or aux-trunk. When the announcement type is integ-mus, this field defaults to b. Callers are connected to the announcement at any time while it is playing. **Note:** The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated circuit pack. The initial request to play an announcement selects an available port on the board on which the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue. Otherwise, the request to play is denied, and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform “calls connected” limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

Queue Length

Use this field to specify the number of calls that can queue for the announcement. The maximum allowed entry depends on the system configuration.

Rate

The recording rate speed in 1000 bits/seconds for TN750 or ISSPA integrated announcements. A different recording speed can be used for each integrated announcement. With VAL type sources, the default is 64 and cannot be changed.

Valid Entry	Usage
16	16 kbps • 8 minutes and 32 seconds of announcement time per circuit pack • 1 hour and 24 minutes for 10 circuit packs for the TN750 • 240 minutes of storage time for the ISSPA This rate does not provide a high-quality recording. Do not use this rate for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps • 4 minutes and 16 seconds of total announcement time for the TN750 • 120 minutes of storage time for the ISSPA
64	64 kbps • 2 minutes and 8 seconds of announcement time per circuit pack • 42 minutes for 10 circuit packs for the TN750 • 60 minutes of storage time for the ISSPA This is the default for VAL

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 11: ARS Toll Table

Use this screen to assign ARS toll tables used by Subnet Trunking. This screen specifies whether calls to the local central office codes listed on the table are toll calls.

To change the value of the fields on the ARS Toll Table screen, use the following command:

change ars toll

Chapter 12: Attendant Console

Use this screen to assign an attendant console to the system.

To add an attendant, use the following command:

`add attendant` *n*, where *n* is the attendant console number.

Attendant console: page 1

Attendant Console x

Number assigned to the attendant console. The attendant console is the main answering position for an organization.

Auto Answer

Use this field to administer the setting of the **auto answer** button on the attendant console.

Valid entry	Usage
all	Enables automatic answering for all incoming calls.
acd	Enables automatic answering only for ACD and direct agent calls.
none	Disables automatic answering.

Console Type

Use this field to specify the type of attendant console.

You can administer only one night-only or day/night type console in the system unless the Tenant Partitioning feature is enabled.

Night Service can be activated from the principal attendant console or the station on which the **nite-serv** button is administered.

Valid entry	Usage
principal	To enable the attendant console to activate Night Service.
day-only	To enable the attendant console to handle only day service calls.
night-only	To enable the attendant console to handle only night service calls.
day/night	To enable the attendant console to handle day or night service calls.

COR

Use this field to assign a class of restriction (COR) number to the attendant console. You must enter a value from 0 to 995.

COS

Use this field to assign a class of service (COS) number to the attendant console. You must enter a value from 0 to 15.

Data Module

Use this field to enable the connection between the attendant console and the data terminal.

Related Links

Data Module on page 178

Disp Client Redir

Use this field to specify whether the attendant console must display the redirection information for a call originating from the station with the Client Room Class of Service feature.

Note:

You must set the **Disp Client Redir** field to `y` for:

- The stations with the audix type.
- The stations with the audix voice power ports or the ports that require the display information.

The system displays this field only if the Hospitality feature is enabled.

Related Links

Hospitality (Basic) on page 800
Hospitality (G3V3 Enhancements) on page 800

Display Language

Use this field to specify the language in which the system must display the console messages. You must enter one of the following values:

- english
- french
- italian
- spanish
- user-defined
- unicode

Note:

The unicode option is available only for unicode-supported telephones, such as 4610SW, 4620SW, 4621SW, 4622SW, 2420J, Sage, Spice, Spark, and 9600-series telephones.

Related Links

Display Character Set on page 787

Extension

Use this field to assign an extension to the attendant console.

Users can call the attendant console by dialing this extension. If an attendant console does not have an extension, the attendant console can only be addressed as a member of the attendant group.

You must not leave this field blank for the attendant console with a data module.

Group

Use this field to assign an attendant group number to the attendant console.

H.320 Conversion

Use this field to enable the conversion of H.320-compliant calls to voice-only calls for the attendant console.

Note:

The system can handle only a limited number of conversion calls. Therefore, the number of attendant consoles with H.320 conversion must be limited.

Name

Use this field to assign a name to the attendant console. You can enter up to 27 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Port

Use this field to assign a port number to the attendant console. For reliability, the attendant consoles must not be assigned to ports on the same digital line circuit pack.

You must enter one of the following values:

- A port number.
- **ip** for a soft attendant console.
- **X** if no hardware is associated with the port assignment.

You can assign the port number using one of the following formats:

- *cabinet number* : *carrier number* : *slot number* : *circuit number*

Valid entry	Usage
01 to 64	The first and second characters represent the cabinet number.
A to E	The third character represents the carrier number.
01 to 20	The fourth and fifth characters represent the slot number.
01 to 31	The sixth and seventh characters represent the circuit number.

- *gateway number* : *module number* : *circuit number*

Valid entry	Usage
001 to 250	The first three characters represent the gateway number.
V1 to V9	The fourth and fifth characters represent the module number.
01 to 31	The sixth and seventh characters represent the circuit number.

Related Links

IP Attendant Consoles on page 800
Extension on page 63

Security Code

The station security code required by the SoftConsole IP attendant.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Type

Valid Entry	Usage
console	The extension is used with any type of console.
302	302B/C/D or SoftConsole IP attendant

Direct Trunk Group Select Button Assignments (Trunk Access Codes)

Use this field to assign the local and remote trunk access codes (TACs) to the attendant console. A local TAC refers to a trunk group or a Loudspeaker Paging zone on the local server. A remote TAC refers to a trunk group on the remote server.

The local TAC can have up to four digits and the remote TAC can have up to three digits. You can use the asterisk (*) and the pound sign (#) as the first digit.

Note:

The local TAC between the local and remote servers must be a DCS trunk. If not , Communication Manager cannot monitor the remote trunk.

Hundreds Select Button Assignments

Use this field to assign a hundreds group to a **Hundreds Select** button on an attendant console. You can enter up to 11 digits.

Use **1** to **8** fields for 24A consoles, and use **1** to **8** fields for 26A consoles. what is this?

Attendant console: page 2 Softconsole IP Attendant

Always Use

Use this field to enable a soft attendant console to register, regardless of the emergency call handling settings on the console. If the user dials `911`, the system uses the administered value in the **Emergency Location Ext** field.

The **Always Use** field is inapplicable to SCCAN wireless consoles and the consoles administered as the H.323 type.

Related Links

Emergency Location Ext on page 716

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

Emergency Location Ext

Use this field to specify the emergency location extension for the soft attendant console. You can enter up to 13 digits. This extension is used as a starting point for identifying the location when an emergency call is made.

The system uses:

- The CAMA Numbering format to modify the extension before sending the extension over the CAMA trunk.
- The Numbering-Public/Unknown format to modify the extension before sending the extension over the ISDN trunk.

IP Audio Hairpinning

If enabled, IP endpoints connect through the IP circuit pack in the server in IP format to bypass the Communication Manager TDM bus.

Remote Softphone Emergency Calls

Use this field to specify how Communication Manager must handle emergency calls from an IP telephone. The system displays this field only when **IP Softphone** is enabled for the system.

Valid entry	Usage
as-on-local	If the administrator populates the IP Address Mapping screen with emergency numbers, the value as-on-local functions as follows: • If the **Emergency Location Extension** field in the Station screen is the same as the **Emergency Location Extension** field in the IP Address Mapping screen, the feature sends the console's own extension number to the Public Safety Answering Point (PSAP). • If the **Emergency Location Extension** field is different from the **Emergency Location Extension** field in the IP Address Mapping screen, the value as-on-local sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).
block	Prevents the completion of emergency calls. Used for users who are mobile but always have a circuit-switched telephone nearby, and for those who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
cesid	Communication Manager can send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Used for IP Softphones with road warrior service that are near enough to the Avaya S8XXX server that an emergency call routed over the trunk reaches the PSAP that covers the server or switch. If the Avaya S8XXX server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the Avaya S8XXX server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Users can select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Used for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. Users choose between block and cesid on the softphone. A DCP or IP telephone in the office automatically selects **extension**.

Related Links

IP Softphone on page 721

Service Link Mode

Use this field to specify the duration of a service link connection. The service link is the combined hardware and software multimedia connection between an H.320 Desktop Video Conferencing (DVC) system and Communication Manager.

The service link is established when a user receives or makes a call during a multimedia, IP softphone, or IP telephone session.

Valid entry	Usage
as-needed	For multimedia, IP softphone, and IP telephone users. The service link remains connected for 10 seconds after the user disconnects a call so that the user can immediately make or receive another call. After 10 seconds, the link is disconnected, and a new link must be established to make or receive a call.
permanent	For call center agents who are constantly making or receiving calls during the multimedia, IP softphone, or IP telephone session. The service link remains connected for the entire duration of the session.

Attendant console: page 2 VIS feature options

VIS FEATURE OPTIONS

These fields administer Visually Impaired Services options.

Auto Start

Use this field to allow or deny permission to an attendant to start a call by pressing any key on the keypad without the need to first press the **Start** button.

Echo Digits Dialed

Enables and disables voiced confirmation of dialed digits.

Attendant console: page 3

Displays if the attendant console is to be connected to a data terminal using a 7400B or 8400 data module.

Related Links

Data Module on page 62

ATTENDANT DATA MODULE

Bcc

Determines compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

COR

Valid Entry	Usage
0 to 995	The Class Of Restriction (COR) number for the data module.

COS

Valid Entry	Usage
0 to 15	The (COS) number used to designate allowed features.

Data Extension

Use this field to assign an extension to the data module. You can enter up to five digits. This value must conform to the dial plan.

Name

The name assigned to the data module extension number.

Related Links

Ext on page 71

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

ABBREVIATED DIALING

List1

Valid Entry	Usage
s	System
g	Group — A group number is also required
p	Personal — A personal list number also is required
e	Enhanced

SPECIAL DIALING OPTION

Use this field to specify the destination of a call when the data module originates the call. Use this field to specify the destination of a call that starts from the data module.

Valid entry	Usage
hot-line	Single-line telephone users can automatically make a call to an extension, a telephone number, or a feature access code.
default	The system dials an Abbreviated Dialing number when the user disconnects.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Abbreviated Dialing Dial Code is the Abbreviated Dialing number dialed when the user goes off-hook on a Data Hotline call.

Using Hotline Service, IP telephone users can place a call to a preassigned destination by going offhook. The preassigned destination can be an extension, a telephone number, or a feature access code. The Hotline Service feature does not work for SIP endpoints.

The Hotline Service destination number is stored in Abbreviated Dialing List.

A Hotline Service destination can be a Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hotline Service when you require very fast service and when you use a telephone only for gaining access to a certain facility. You can use Loudspeaker Paging Access with Hotline Service to provide automatic access to paging equipment.

This field is available only for a default or hotline special dialing option.

Related Links

SPECIAL DIALING OPTION on page 70

DEFAULT DIALING Abbreviated Dialing Dial Code

The AD number dialed when the user goes off-hook and enters a carriage return following the “DIAL” prompt. The data call originator also can perform data terminal dialing by specifying a dial string that might or might not contain alphanumeric names. Available only for a default special dialing option.

Related Links

SPECIAL DIALING OPTION on page 70

ASSIGNED MEMBER

Ext

The extension number of a previously-administered user who has an associated **Data Extension** button and who shares the use of the module.

Name

The name assigned to the data module extension number.

Related Links

Ext on page 71

Attendant console: page 3 feature button assignments

FEATURE BUTTON ASSIGNMENTS

Administers the feature buttons assigned to the attendant console. The split and forced release buttons are administered in a fixed location. The **hold**, **night-serv**, and **pos-busy** buttons have default locations. The following table provides descriptions of feature buttons that are unique to the attendant console.

Audible Tones On/Off

Valid Entry	Usage
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).

Attendant Control of Trunk Group Access

Valid Entry	Usage
act-tr-grp	Activate trunk group access. The attendant can control a trunk group. All calls going to the trunks are routed to the attendant (one per console).
deact-tr-g	Deactivate trunk group access. The attendant can release control of a trunk group (one per console).

Table continues…

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Valid Entry	Usage
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (one per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there are one or more calls on the emergency attendant queue (one per console).
hold	Hold. When the Hold button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the call type button associated with the loop is lit (one per console).
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the "Pos Avail" light is turned off, and the light associated with the **pos-busy** button is lit. Pushing the **pos-busy** button a second time takes the console out of “position busy” mode, turns on the “Pos Avail” light and turns off the light associated with the **pos-busy** button. If the **pos-busy** button is administered on a 2-LED button, the top LED flashes when the last attendant goes into “Position Busy” mode. Otherwise, if the button has only one LED, the single LED associated with the **pos-busy** button flashes (one per console).
serial-cal	Serial Call. The attendant-extended calls can return to the same attendant if the trunk remains off-hook (one per console).
override	Attendant Override. The attendant can override diversion features such as, Call Forwarding, Call Coverage, and so on (one per console).
intrusion	Call Offer. The attendant can extend a call when the called party is active on another call (one per console).
dont-split	Don’t Split. The attendant cannot split away a call when dialing (one per console).
vis	Visually Impaired Attendant Service (vis). Activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: • “con-stat” repeats the console status. • “display” calls out display contents. • “dtgs-stat” calls out the DTGS status. • “last-mess” repeats the last message. • “last-op” calls out the last operation.

Trunk Group Select

Up to 12 **DTGS** buttons can be administered. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using **Attendant Control of Trunk Group Select** buttons.

Valid Entry	Usage
local-tgs	Local trunk group select. The attendant can access trunk groups on the local server running Communication Manager.
remote-tgs	Remote trunk group select. The attendant can access trunk groups on a remote server running Avaya Communication Manager.

Other

Valid Entry	Usage
alt-frl	Alternate facility restriction level. The attendant can activate or deactivate the AFRL feature. When activated, the originating device (lines or trunks) can use an alternate set of the facility restriction levels to originate a call (one per console).
hundrd-sel	Hundreds group select. Additional administered hundreds group select feature buttons. When a feature button is administered as “hundrd-sel”, the system displays a subfield that must be administered with a 1 to 3 digit hundreds group plus prefix, if needed. Administered hundrd-sel feature buttons operate in the same manner as fixed **HGS** buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Therefore, if all 20 fixed **HGS** buttons have been administered, no additional **hundrd-sel** feature buttons can be administered. If 12 HGS buttons are assigned, reassign the **night**, **pos-busy**, and **hold** buttons to locations 20, 21, and 3, respectively. The **HGS** buttons should then be assigned to the right-most three columns, as required.
group-disp	Group Display. The attendant can see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. The attendant can select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.
occ-rooms	Occupied rooms. The attendant can see which rooms are occupied.
maid-stat	Maid status. The attendant can see which rooms are in one of six specified states.
vu-display	VuStats (**vu-display**). Users with display telephones and attendants can turn on the VuStats display. The limit to the number of VuStats feature buttons depends on how many feature buttons are available on the attendant console. You can set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.

Attendant console: page 4 display module button assignments

Display Module Button Assignments

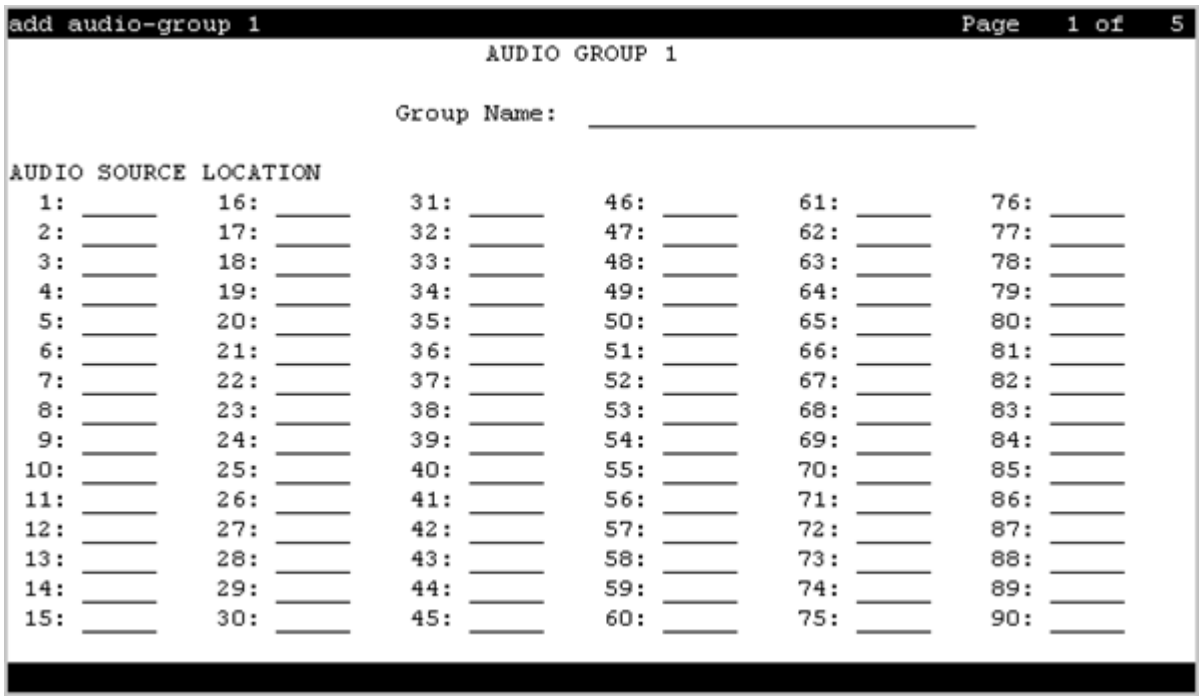
Display-type buttons obtain display functions on the associated alphanumeric display. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is inoperative. These buttons are noted as [status lamp]. If a **Call Cover Msg Rt** (cov-msg-rt) button is assigned, a Leave Word Calling Delete Msg (delete-msg) button and a **Next** (next) button must also be assigned.

Chapter 13: Audio Group

Use this screen to assign an audio group to the system. The audio group is a collection of recorded audio sources that have been placed in the group to facilitate the selection.

To add an audio group, use the following command:

add audio-group *n*, where *n* is the audio group number.



Audio Source Location

Use this field to assign a board location to the audio group. You must enter the value using one of the following formats:

- *aaxss*, where *aa* is a cabinet number, *x* is a carrier number, and *ss* is a slot number.
- *ggg*v*x*, where *ggg* is a gateway number and v*x* is the module number.

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Group Name

Use this field to assign an alphanumeric name to the audio group.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Chapter 14: AUDIX-MSA Node Names

Example command: `change node-names audix`

AUDIX Names

The name of the voice messaging node consisting of a one- to seven-character string. Used as a label for the associated IP address. The node names must be unique on each server running Communication Manager.

IP Address

The IP address associated with the node name.

MSA Names

Identifies the name of the MSA node. Accepts a one- to seven-character string. The MSA names must be unique on each server running Communication Manager.

Chapter 15: Authorization Code — COR Mapping

Use this screen to administer an authorization code and the class of restriction (COR) number associated with the authorization code.

Security alert:

To maximize security, you must:

- Administer authorization codes to the acceptable maximum length.
- Create random authorization codes.
- Change authorization codes at least quarterly.
- Deactivate authorization codes immediately if a user leaves the company or changes assignments.
- Assign authorization codes with the minimum level of calling permission.

To change the value of the fields on the Authorization Code — COR Mapping screen, use the following command:

`change authorization-code` *n*, where *n* is the authorization code.

AC

Use this field to administer an authorization code (AC). The AC can be any combination of numbers and can consist of 4 to 13 digits. To enhance the system security, you must administer the AC of 13 random digits.

Related Links

Authorization Code Length on page 283

COR

Use this field to assign a class of restriction (COR) number to the authorization code. You can enter a value from 0 to 995.

Number of Codes Administered

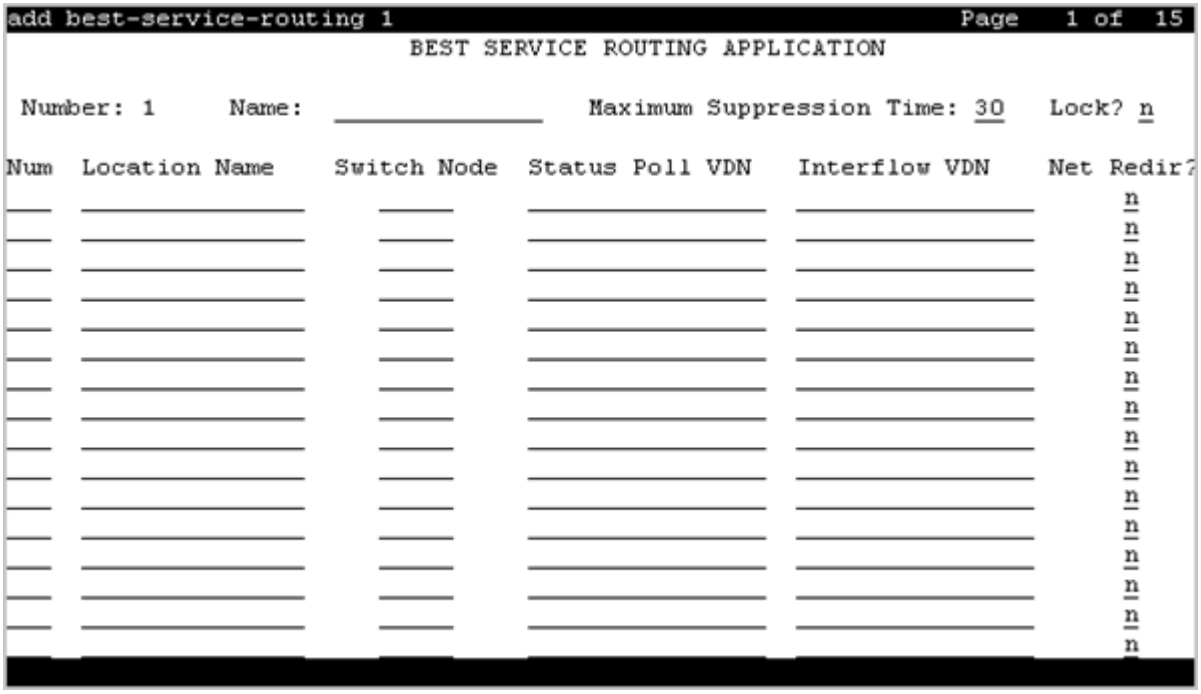
The number of authorization codes already administered. There is a maximum number of authorization codes depending on system configuration.

Chapter 16: Best Service Routing

Use this screen to enable the system to compare the specified skills, identify the skill that provides best service to a call, and deliver the call to an agent in the skill. If no agents are available in the skill, the call is queued. You can also use this screen to monitor the status of the specified skills and adjust call processing and routing as appropriate.

To change the value of the fields on the Best Service Routing screen, use the following command:

`change best-service-routing` *n*, where *n* is the routing number



Interflow VDN

The routing number including the dial access code used to access the Interflow Vector Directory Number (VDN) at the remote location. Accepts up to 16 characters.

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Valid entry	Usage
0 to 9, *, #	To enter the digit.
~p	To enable the system to pause.
~w or ~W	To enable the system to wait.
~m	To enable the system to mark.
~s	To enable the system to suppress.

Location Name

A name that identifies each location where this feature is administered. Accepts up to 15 alphanumeric characters.

Lock

Valid Entry	Usage
y	Provides extra security by not sending any BSR information to the Call Management System.
n	Sends BSR information to the CMS.

Maximum Suppression Time

Use this field to prevent callers from connecting to a VDN during the specified period after receiving a busy signal.

Valid entry	Usage
0 to 60	The maximum poll suppression time in seconds.

Net Redir

Enables or disables Network Call Redirection.

Num

Use this field to specify a location number. You can enter a value from 1 to 255.

Number

Valid Entry	Usage
1 to 255	The identifying number of the current BSR plan.

Status Poll VDN

The routing number including the dial access code used to access the Status Poll VDN at the remote location. Accepts up to 16 characters.

Valid entry	Usage
0 to 9, *, #	To enter the digit.
~p	To enable the system to pause.
~w or ~W	To enable the system to wait.
~m	To enable the system to mark.
~s	To enable the system to suppress.

Switch Node

Valid Entry	Usage
1 to 32767 blank	The Network Node ID for each switch if the Universal Call ID is being used. This is optional.

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Chapter 17: Bulletin Board

The bulletin board is used to post and receive information. The first 10 lines are for high-priority messages from Avaya personnel. An init or inads login is used to enter high-priority information to trigger the high-priority message at login time. Additional lines can be used by anyone with access. The system automatically enters the date the message was posted or last changed to the right of each message line.

Accommodates up to 40 characters of text per line and one blank line. If more than one blank line is entered, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. Text cannot be indented on the bulletin board.

Example command: `change bulletin-board`

Date

The date the bulletin board information was entered or last changed.

Text lines

Used for high priority messages on the bulletin board. Anyone with an init or inads login can enter high-priority information to trigger the high-priority message at login time. Additional lines can be used by anyone with access.

Chapter 18: Button Type Customization Restriction

Restricts customized button labels of up to 50 specified button types for users with normal access. Manages system allocation of customized button labels to ensure that VIP users have the available customized button label resources.

Example command: `change button restriction`

Restrict Customization Of Button Types

Valid Entry	Usage
y	Restricts the use of customized feature button labels. This is the default.
n	Users can customize labels for all buttons on their telephones without any restrictions.

Restrict Customization Of Labels For the Following Button Types

The button type restricted from being customized.

Note:

The **abr-spchar** and the **abrv-dial** button types require a special associated character.

- abrv-dial

  An abbreviated dialing list type: e = enhanced, g = group, p = personal, and s = system

- abr-spchar

  A ~-character in an abbreviated dialing digit string. ~, ~m, ~p, ~s, ~w, and ~W. For more information on digit strings, see Dial Code.

Available only when **Restrict Customization of Button Types** is enabled.

Related Links

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Restrict Customization Of Button Types on page 83
DIAL CODE on page 33

Chapter 19: Call Type Digit Analysis Table

Specifies how to modify telephone numbers for internal contacts dialed from the telephone call log or from a corporate directory. Users can automatically place outgoing calls based on the telephone number information in the telephone call log without having to modify the telephone number. Requires at least one entry to activate.

Example command: `change calltype analysis`

Delete

Communication Manager deletes this number of digits in the original digit string from the left-hand side of the original digit string to complete analysis and routing.

Dialed String length (Min, Max)

Communication Manager compares digit strings of this length to the original digit string, looking for a match to complete analysis and routing.

Dialed String Match

Communication Manager compares this digit string to the original digit string, looking for a match to complete analysis and routing. The characters x and X can be used as wildcards.

Insert

Communication Manager inserts these digits into the left-hand side of the original digit string to complete analysis and routing.

Location

Valid Entry	Usage
Numeric value all	Telephones dialing from this location use the entries in the Call Type Digit Analysis Table. If there are matching entries in the telephone's location, those entries are used. If there are no matching entries in the telephone's location, the Communication Manager tries the entries in location all.

Type

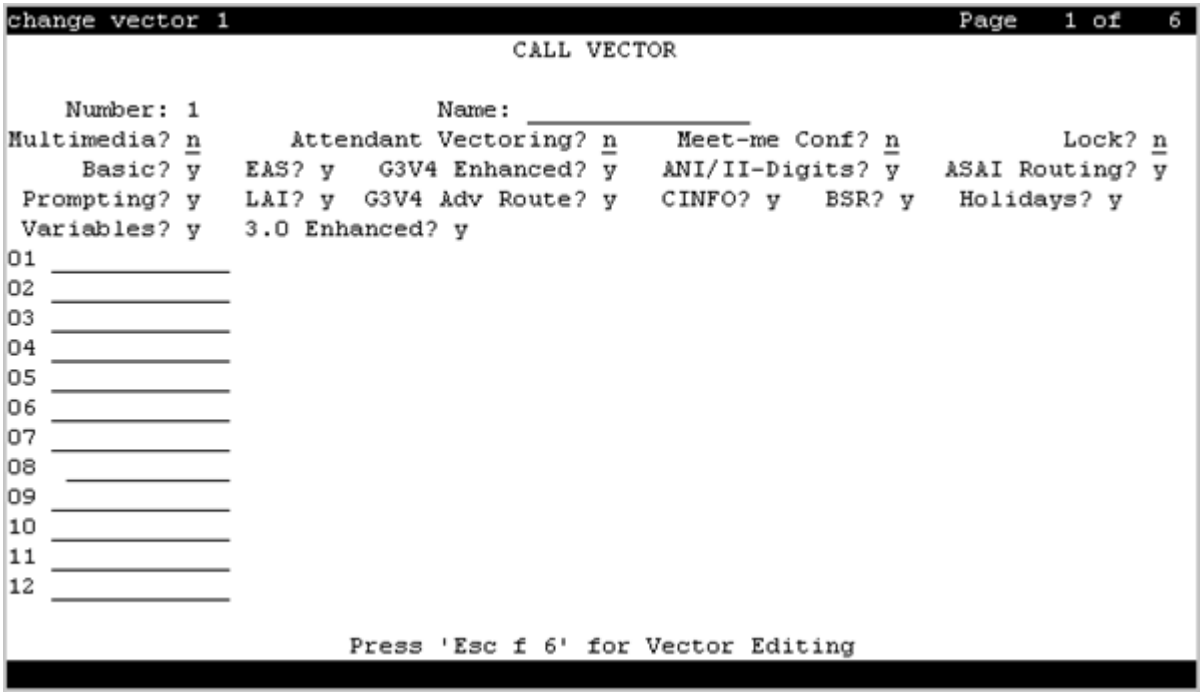
The administered call type for this dialed string. Communication Manager tests the modified digit string against the administered call type.

Valid Entry	Usage
aar	Automatic Alternate Routing — digit analysis algorithm commonly used for private network calls.
ars	Automatic Route Selection — digit analysis algorithm commonly used for public network calls.
ext	Extension entries in the dialplan analysis tables of type ext.
udp	Extension entries in the **uniform-dialplan tables**.

Chapter 20: Call Vector

Programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN).

Example command: `change vector n`, where *n* is the vector number.



01 through XX

The following vector commands that specify how to handle calls directed to a VDN are located on lines 01 through *XX*. The maximum allowed depends on the configuration.

Valid Entry	Usage
adjunct routing	Causes a message to be sent to an adjunct requesting routing instructions based on the CTI link number.
announcement	Provides the caller with a recorded announcement.

Table continues…

Valid Entry	Usage
busy	Gives the caller a busy signal and causes termination of vector processing.
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).
collect	The user can enter up to 16 digits from a touch-tone telephone, or the vector can retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the server running Communication Manager. Also, the optional assignment of an announcement plays immediately before the call is disconnected.
goto	Used for conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	The caller can leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it "returns" best data for its location to the primary vector on the origin server.
return	Returns vector processing to the step following the `goto` command after a subroutine call has processed.
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number). Communication Manager prepends a plus (+) digit before the called number when invoking Network Call Redirection (NCR).
set	Performs arithmetic and string operations and assigns values to a vector variable or to the digits buffer during vector processing.
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command's syntax. Also provides feedback (in the form of silence, ringback, or music) to the caller while the call advances in queue.

ANI/II-Digits

Indicates whether the ANI and II-Digits vector routing commands can be used or not. ANI/II-Digits Routing is a **G3V4 Enhanced** feature.

Related Links

G3V4 Enhanced on page 90

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ASAI Routing

Indicates whether or not the **CallVisor Adjunct/Switch Applications Interface (ASAI) Routing** option is active or inactive.

Related Links

Attendant Vectoring on page 796

Attendant Vectoring

Indicates whether or not Attendant Vectoring is selected on this VDN. Attendant Vectoring does not support Call Center features.

Related Links

Attendant Vectoring on page 796

Basic

Indicates whether Vectoring (Basic) is enabled for system.

Related Links

Vectoring (Basic) on page 810

BSR

Indicates if the Best Service Routing option is enabled on the system. BSR commands and command elements can be used in vectors only if this option is enabled.

Related Links

Vectoring (Best Service Routing) on page 811

CINFO

Indicates whether Caller Information Forwarding (CINFO) Routing is enabled on the system. A call with CINFO routes based on the digits supplied by the network in an ISDN-PRI message.

Related Links

Vectoring (CINFO) on page 811

EAS

Indicates if Expert Agent Selection (EAS) is enabled on the system.

Note:

When EAS is enabled, the terminology changes from “Split” to “Skill” for help messages, error messages, and vector commands. For example, check backup split becomes check backup skill.

Related Links

Expert Agent Selection (EAS) on page 808

G3V4 Adv Route

Indicates whether G3V4 Advanced Vector Routing commands are allowed.

Related Links

Vectoring (G3V4 Advanced Routing) on page 811

G3V4 Enhanced

Indicates whether G3V4 Enhanced Vector Routing commands and features are allowed.

Related Links

Vectoring (G3V4 Enhanced) on page 811

Holidays

Indicates whether Holiday Vectoring features are allowed.

Related Links

Vectoring (Holidays) on page 811

LAI

Indicates if Look-Ahead Interflow is enabled.

Related Links

Lookahead Interflow (LAI) on page 809

Lock

Controls access to the vector from the Call Management System.

Security alert:

Always lock vectors that contain secure information (for example, access codes).

Valid Entry	Usage
y	This vector is inaccessible from the CMS. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If Meet-me Conference is enabled, **Lock** must also be enabled.
n	Gives CMS users the ability to administer this vector.

Meet-me Conf

Indicates if the Meet-me Conference feature is enabled. If enabled, designates the VDN as a Meet-me Conference VDN. Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

Related Links

Enhanced Conferencing on page 799

Multimedia

Valid Entry	Usage
y	Multimedia calls are set to be answered and billed at the start of vector processing. Multimedia Call Handling must be enabled.
n	The vector stops receiving multimedia calls. Multimedia Call Handling is disabled. This is the default.

Related Links

Multimedia Call Handling (Basic) on page 803
Multimedia Call Handling (Enhanced) on page 803

Name

Optionally provides a reference for the vector name. Used to activate Network Call Redirection. Accepts up to 27 alphanumeric characters.

Available only if ISDN Network Call Redirection is enabled.

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

ISDN/SIP Network Call Redirection on page 802
Display Character Set on page 787

Number

The vector number.

Prompting

Indicates whether Vectoring (Prompting) is enabled for the system.

Related Links

Vectoring (Prompting) on page 811

Chapter 21: Calling Party Number Conversion for Tandem Calls

Use this screen to administer Calling Party Number formats for tandem calls. Tandem calls are the calls that were not originated locally on the switch.

Tandem calls to the public network may not provide the correct calling party information. To rectify this problem, Communication Manager changes Calling Party Number. To generate the correct Calling Party Number, Communication Manager compares the incoming calling party number with the calling party digit lengths, calling party pefixes, and trunk groups. When a match is found, Communication Manager constructs a Calling Party Number by deleting the digits specified in the **Delete** field, and then inserting the digits specified in the Insert field. To construct the values of Numbering Plan Identifier (NPI) and Type of Number (TON) for Calling Party Number Communication Manager uses the numbering format specified in the Number Format field.

Note:

Communication Manager does not update Calling Party Number in the NCA-TSC SETUP message.

To map numbers for ISDN-SIP interworking, Communication Manager uses the values of the fields on the Calling Party Number Conversion for Tandem Calls screen or sends the call to Session Manager. Session Manager uses the number adaptation module for the mapping.

Note:

During ISDN to SIP mapping, the number adaptation module does not have access to NPI and TON values in the incoming ISDN number. During SIP to ISDN mapping, Communication Manager uses the leading plus (+) sign to process an incoming SIP number as an international number and insert the plus (+) sign in the resulting number. To process an incoming SIP number, Communication Manager uses the values of the fields on the Calling Party Number Conversion for Tandem Calls screen.

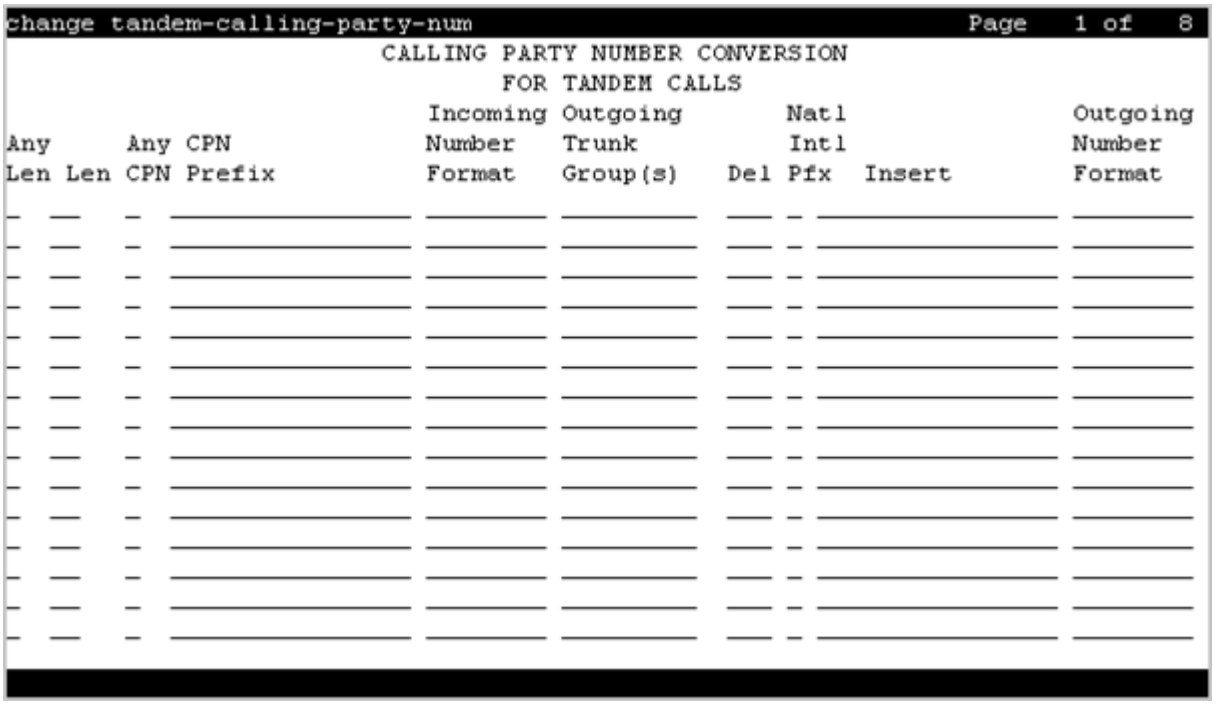
The following rules apply to an incoming calling party number with a leading plus (+) sign:

- For an incoming SIP calling party number with a leading plus (+) sign, Communication Manager includes the plus (+) sign when searching for a match.
- For an incoming SIP calling party number with a leading plus (+) sign, if a match is not found with the plus (+) sign, Communication Manager removes the plus (+) sign.
- For an incoming ISDN calling party number reconstructed as an international number, Communication Manager searches for a match. If a match is not found, Communication Manager adds a leading plus (+) sign.

For an incoming SIP calling party number, if Communication Manager finds a match, the matching entry overrides the setting of the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field for the SIP signaling group.

To modify the value of the fields on the Calling Party Number Conversion for Tandem Calls screen, use the following command:

```
change tandem-calling-party-num
```



Related Links

Delete on page 95
Insert on page 96
Incoming Number Format on page 96
Modify Tandem Calling Number on page 501

Any CPN

Valid entry	Usage
x	When checking for a matching entry, Communication Manager does not consider the digits in the calling-party number. Communication Manager disables the **CPN prefix** field.
blank	When checking for a matching entry, Communication Manager compares the digits in calling-party number with the digits entered in the **CPN prefix** field. By default, this field is blank.

Any Len

Valid entry	Usage
x	During conversion, Communication Manager does not check the length of the calling-party number. Communication Manager disables the **Len** field.
blank	During conversion, Communication Manager checks the length of the calling-party number. Communication Manager matches the length of the calling-party number with the length that is entered in the **Len** field. By default, this field is blank.

CPN Len

Valid Entry	Usage
1 to 15	A number between 1 and 15 that indicates the length of calling party number (CPN).
blank	Leave blank when deleting an entry. This is the default value.

CPN Prefix

Any combination of digits 0 to 9, up to 16 digits. The '+' sign is also valid. Indicates that all trunk groups are valid provided the **Modify Tandem Calling Party** field on the ISDN Trunk Group screen is set to y. A specific CPN digit string match is not required, provided other matching criteria for tandem CPN modification are met. This is the default value. You can enter up to 16 digits to indicate the calling party prefix.

Delete

Valid Entry	Usage
1 to 15	The number of digits deleted when modifying the tandem calling party number.
all	All digits are deleted.
blank	No digits are deleted from the received calling party number. This is the default value.

Insert

Valid Entry	Usage
1 to 15	The number of digits inserted when modifying the tandem calling party number.
blank	The received calling party number is not prefixed with any digits. This is the default value.

Incoming Number Format

Valid Entry	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	The numbering format to use in modifying the tandem calling party number.
blank	All numbering formats match this entry.

Natl Intl Pfx

Valid entry	Usage
n(atl)	Communication Manager adds the **Long distance access code** number as the prefix to the calling-party number. The **Long distance access code** field is on page 1 of the Location parameter screen. If the **Long distance access code** field is blank, Communication Manager checks the prefix number in the **National CPN Prefix** field on the Feature related system parameter screen.
i(ntl)	Communication Manager adds the **International access code** number as the prefix to the calling-party number. The **International access code** field is on page 1 of the Location parameter screen. If the **International access code** field is blank, Communication Manager checks the prefix number in the **International CPN Prefix** field on the Feature related system parameter screen.

Outgoing Trunk Group(s)

Valid entry	Usage
Trunk group number	Enter an ISDN or a SIP outgoing trunk group number.
blank	If you leave this field blank, all trunk groups are valid only if you enable the **Modify Tandem Calling Number** field for the ISDN or SIP trunk group. By default, this field is blank.

Chapter 22: CAMA Numbering Format

Administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

Provides the CESID format by extension number or number blocks. Multiple CESID formats can be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

Example command: `change cama-numbering`

CESID

The number used to identify the calling terminal within an emergency service system. This field can represent a prefix to an extension or the entire CESID. Accepts up to 16 digits. The maximum number of allowed digits depends on the equipment.

Ext Code

The leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code is interpreted as a block of digits.

Example: If the extension length is 4 and the extension code is 11, the CESID serves extensions 1100 through 1199.

Ext Len

The number of digits in the extension. Accepts up to 13 digits.

System CESID Default

The number sent over the CAMA trunk if the Extension Code for the CESID is unspecified. Accepts up to 16 digits.

Related Links

Ext Code on page 98

Total Length

The total number of sent digits. Accepts up to 16 digits.

Chapter 23: CDR system parameters

Administers the Call Detail Recording (CDR) feature used to record information on incoming, outgoing, and tandem calls for each trunk group administered for CDR, including auxiliary trunks. The system records information on each trunk-group call and each station-to-station call.

Example command: `change system-parameters cdr`

CDR system parameters: page 1

Calls to Hunt Group — Record

Valid Entry	Usage
member-ext	Records the extension of the telephone or data terminal where the call terminated.
group-ext	Records the extension that was dialed.

CDR Account Code Length

Valid Entry	Usage
1 to 15	The number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

CDR Date Format

Valid Entry	Usage
month/day day/month	The format for the date stamp that begins each new day of call records.

Condition Code ‘T’ for Redirected Calls

Enables or disables identification of CDR records of calls that have been redirected automatically off the server running Communication Manager.

Valid Entry	Usage
y	The Condition Code of both CDR records for the call is ‘T.’
n	Special identification is applied.

Digits to Record for Outgoing Calls

Valid Entry	Usage
dialed	Record the digits a user dials.
outpulsed	Record the digits that Communication Manager sends out over the trunk, including any additions or deletions that take place during routing.

Disconnect Information in Place of FRL

Valid Entry	Usage
y	Replace the Facility Restriction Level (FRL) with information about why a call disconnects.
n	Record the call FRL.

Enable CDR Storage on Disk

Enables or disables the Survivable CDR feature for the main server, Survivable Remote Server (Local Survivable Processor), and Survivable Core Server (Enterprise Survivable Server). Default is disabled.

Force Entry of Acct Code for Calls Marked on Toll Analysis Form

Specifies whether or not an account code is required when making a toll call. Account codes are not required on all charged calls, and are required on some non-charged calls.

Valid Entry	Usage
y	Restricts all toll calls unless the user dials an account code. Available only if Forced Entry of Account Codes is enabled for the system.

Table continues…

Comments on this document? infodev@avaya.com

Valid Entry	Usage
n	Users can place a call without an account code. This does not override other calling restrictions.

Related Links

Forced Entry of Account Codes on page 800

Inc Attd Call Record

Enables or disables separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Available only if Incoming Trunk Call Splitting is enabled.

Related Links

Inc Trk Call Splitting on page 102

Inc Trk Call Splitting

Enables or disables the creation of separate records for each portion of incoming calls that are transferred or conferenced.

Available only if **Record Outgoing Calls Only** is disabled.

Related Links

Record Outgoing Calls Only on page 106

Interworking Feat-flag

Valid Entry	Usage
y	The feature flag indicates interworked outgoing ISDN calls.
n	The feature flag indicates no answer supervision for interworked calls.

Intra-Switch CDR

Enables or disables the recording of calls within Communication Manager. Requires administration of intra-switch CDR extensions.

Related Links

Intra-Switch CDR on page 410

Modified Circuit ID Display

Affects the “printer”, “teleseer”, and “59-character” output formats.

Valid Entry	Usage
y	Displays the circuit ID in its actual format (100’s, 10’s, units). For example, circuit ID 123 displays as 123. **Note:** Requires verification that an output device can accept this format.
n	Displays the circuit ID in its default format (10’s, units, 100’s). For example, the system displays circuit ID 123 as 231.

Node Number (Local PBX ID)

Displays the DCS switch node number in a network of switches.

Outg Attd Call Record

Enables or disables separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Available only if Outgoing Trunk Call Splitting is enabled.

Related Links

Inc Trk Call Splitting on page 102
Outg Trk Call Splitting on page 103

Outg Trk Call Splitting

Enables or disables the creation of separate records for each portion of outgoing calls that are transferred or conferenced

Primary Output Endpoint

Determines where the server running Communication Manager sends the CDR records. Required if a Primary Output Format is specified.

Valid Entry	Usage
eia	The EIA port is used to connect the CDR device.
Extension number	The extension of the data module that links the primary output device to the server running Communication Manager.
CDR1, CDR2	The CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 for IP Services.

Related Links

IP Services on page 473

Primary Output Format

Indicates the format of the call records sent to the primary output device.

Valid Entry	Usage
customized	For special call accounting needs that are prevented for standard record formats. Requires call accounting software that is also customized to receive these records. Call accounting vendors should be consulted before using this option.
printer	Call detail records are sent to a printer rather than to a record collection or call accounting system.
59-char expanded lsu lsu-expand int-direct int-isdn int-process teleseer unformatted	For standard record formats. The selection must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Privacy — Digits to Hide

Indicates how much of the dialed number to hide on the CDR record for stations administered for CDR privacy.

Valid Entry	Usage
0 to 7	The number of digits to hide, counting from the end right to left.

Example

For a value of 4, when the user dials 555-1234, the system displays only “555” in the CDR record.

Related Links

CDR Privacy on page 710

Record Agent ID on Incoming

Determines whether the EAS agent login ID mut be included instead of the physical extension in the CDR record.

Available only if Expert Agent Selection (EAS) is enabled for the system. Cannot be enabled if **Record Called Vector Directory Number Instead of Group or Member** is enabled.

Related Links

Record Called Vector Directory Number Instead of Group or Member on page 105

Record Agent ID on Outgoing

Determines whether the EAS agent's LoginID must be included instead of the physical extension in the CDR record.

Available only if Expert Agent Selection (EAS) is enabled for the system.

Related Links

Expert Agent Selection (EAS) on page 808

Record Call-Assoc TSC

Enables or disables the creation of records for call-associated temporary signaling connections. A large number of data connections could increase the number of records, so call collection device capacity must be taken into consideration.

Record Called Vector Directory Number Instead of Group or Member

Determines whether or not to include the Vector Directory Number (VDN) in the CDR record. If enabled, the called VDN overrides the group or member information that normally the system displays in the CDR record. If a call is directed through more than one VDN, the first VDN used for the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

Cannot be enabled if **Record Agent ID on Incoming** is enabled.

Related Links

Record Agent ID on Incoming on page 105

Record Non-Call-Assoc TSC

Enables or disables the creation of records for non-call-associated temporary signaling connections. A large number of data connections could increase the number of records, so record collection device capacity must be taken into consideration.

Record Outgoing Calls Only

Valid Entry	Usage
y	Record only outgoing calls. **Note:** This can save space if you are only concerned with charges for outbound calls.
n	Record both outgoing and incoming calls.

Remove '+' from SIP Numbers

If you set the **Remove '+' from SIP Numbers** field on the CDR System Parameters screen to y and if a leading plus (+) sign is present in a calling or called number sent over a CDR adjunct, Communication Manager removes the leading plus (+) sign from the number. The default value is y.

Remove # From Called Number

Valid Entry	Usage
y	Removes the “#” or “E” symbol from the **Dialed Number** field of the call detail record. The output device must be able to accept this format.
n	Retains the trailing “#” or “E” symbol in the **Dialed Number** field whenever an inter-digit time-out occurs or users dial # to indicate the end of dialing.

Secondary Output Endpoint

Available only if **Secondary Output Format** is administered.

Valid Entry	Usage
eia	The secondary output device is connected to the eia port.
Extension number	The extension of the data module that links the secondary output device to the server running Communication Manager.

Table continues…

Valid Entry	Usage
CDR1, CDR2	The CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 for IP Services.

Related Links

Secondary Output Format on page 107
Service Type on page 474

Secondary Output Format

Caution:

Only qualified Avaya service personnel should administer a secondary output device. This option might cause loss of data when the buffer contains large amounts of data.

Valid Entry	Usage
customized int-direct int-process lsu unformatted	The format of the call records sent to the secondary output device. Only these formats can be used for a secondary output device. The format must be compatible with call accounting software. Verify this through the vendor or the accounting system documentation.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are the calls that are blocked because the user has insufficient calling privileges or all outgoing trunks are busy. These calls appear on the CDR record with a condition code “E”.

Valid Entry	Usage
y	Ignores ineffective call attempts. Used when there is limited storage space for CDR records and when records often overrun the buffer.
n	Reports ineffective call attempts. Used when users are often unable to place outgoing calls, or if a large number of incoming calls are incomplete. Also used to keep records of attempts to contact a client, and when ISDN trunks are used. This option requires more space for records.

Use Enhanced Formats

Enables or disables the use of the Enhanced version of the specified primary output formats. Enhanced formats provide additional information about time in queue and ISDN call charges, where

available. This affects the expanded, teleseer, lsu, printer, and unformatted output formats. Enhanced formats and ISDN formats cannot be used at the same time.

Use ISDN Layouts

Enables or disables using the ISDN version of the specified primary output format. ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects lsu and printer output formats, as well as any format with ISDN layouts, such as teleseer. ISDN formats and Enhanced formats cannot be used at the same time.

Use Legacy CDR Formats

Enables or disables the use of pre-Communication Manager 4.0 legacy Call Detail Recording (CDR) formats for CDR records. The default is enabled. Listed below are the CDR formats that are impacted by this field. All other CDR formats remain unchanged.

CDR Format	Communication Manager 3.1 and earlier length	Communication Manager 4.0 and later length
ISDN Teleseer	80	82
Enhanced Teleseer	81	83
ISDN Printer	84	86
Enhanced Printer	85	87
ISDN LSU	59	61
Enhanced LSU	59	61
Expanded	135	139
Enhanced Expanded	151	155
Unformatted	105	109
Enhanced Unformatted	119	123
Int-ISDN	136	140

CDR System Parameters: page 2

Used if there is an arrangement with the vendor to customize the call accounting system to receive these records.

Available only if **Primary Output Format** is customized.

Related Links

Primary Output Format on page 104

Data Item

Itemizes the data items that appear on the customized record.

At least one field for a record must be included. The last two data items in the record must be line-feed and return, in that order.

Data Item	Length
acct-code	15
attd-console	2
auth-code	7
bandwidth	2
bcc	1
calltype	1
calling-num	15
clg-pty-cat	2
clg-num/in-tac	10
code-dial	4
code-used	4
cond-code	1
contact-uri	20
country-from	3
country-to	3
date	6
dialed-num	23
duration	4
end-date(4d)	8
feat-flag	1
frl	1
from-uri	20
in-crt-id	3
in-trk-code	4
ins	3
internal-codec	2
isdn-cc	11
isdn-cc/ppm	11
ixc-code	4

Table continues…

Data Item	Length
line-feed	1
local-time-from	
local-time-to	
location-from	3
location-to	3
ma-uui	1
node-num	2
null	3
out-crt-id	3
ppm	5
request-uri	20
res_flag	1
return	1
sec-dur	5
seq-num	10
space	1
start-date	6
start-date(4d)	8
start-time	6
time	4
timezone-from	3
timezone-to	6
to-uri	20
tsc_ct	4
tsc_flag	1
trunk-codec	2
ucid	20
vdn	5

Length

The length of each data item. The maximum record length depends on the call accounting system. Check with the vendor. The date field should be six digits to ensure proper output. Certain fields default to the required length.

Record Length

Displays the accumulated total length of the customized record, updated each time the length of a data item changes.

Chapter 24: Change Station Extension

Changes extensions on the switch from one extension to another all at once. Specifically:

- All administration that was associated with the current extension is associated with the new extension.
- Any administration references on the changing extension, such as references used in a vector, coverage, and so on, are now referenced to the new extension.
- All references to the previous extension are removed from the system.

If an extension is changed that is also administered on an adjunct, such as voice mail or an ASAI link, the extension on the adjunct must also be changed to ensure proper functionality.

For fields that are changing, current information prior to the change displays under **From Extension**.

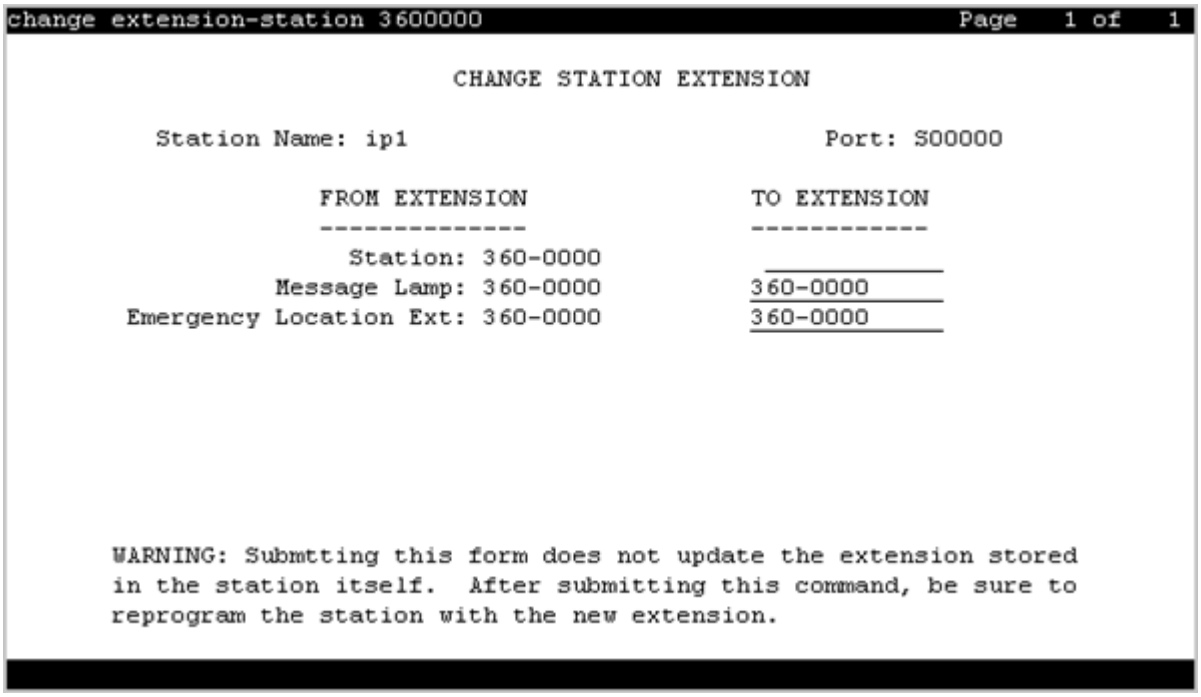
A forwarded extension administered as a button remains.

Exceptions

Station extensions cannot be changed if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- First assign station B to a different emergency location extension.
- First change station B. The system displays station A's extension as the emergency location extension.

Example command: `change extension-station` *n*, where *n* is the extension assigned to a station.



Emergency Location Extension

The Station screen displays the new extension for the **Emergency Location Ext.** field. Up to thirteen numbers can be used to make up a valid extension number for the dial plan.

IP Parameter Emergency Location

The emergency location extension from the IP Address Mapping screen associated with the current extension.

Related Links

[IP network region](#) on page 436
[Emergency Location Extension](#) on page 411

Message Lamp

The new extension for the message lamp extension. Up to thirteen numbers can make up a valid extension number for the dial plan.

Port

The port of the existing extension.

Station

The new extension. Up to seven numbers can make up a valid extension number for the dial plan.

Station Name

The name of the existing extension.

Chapter 25: Circuit Packs

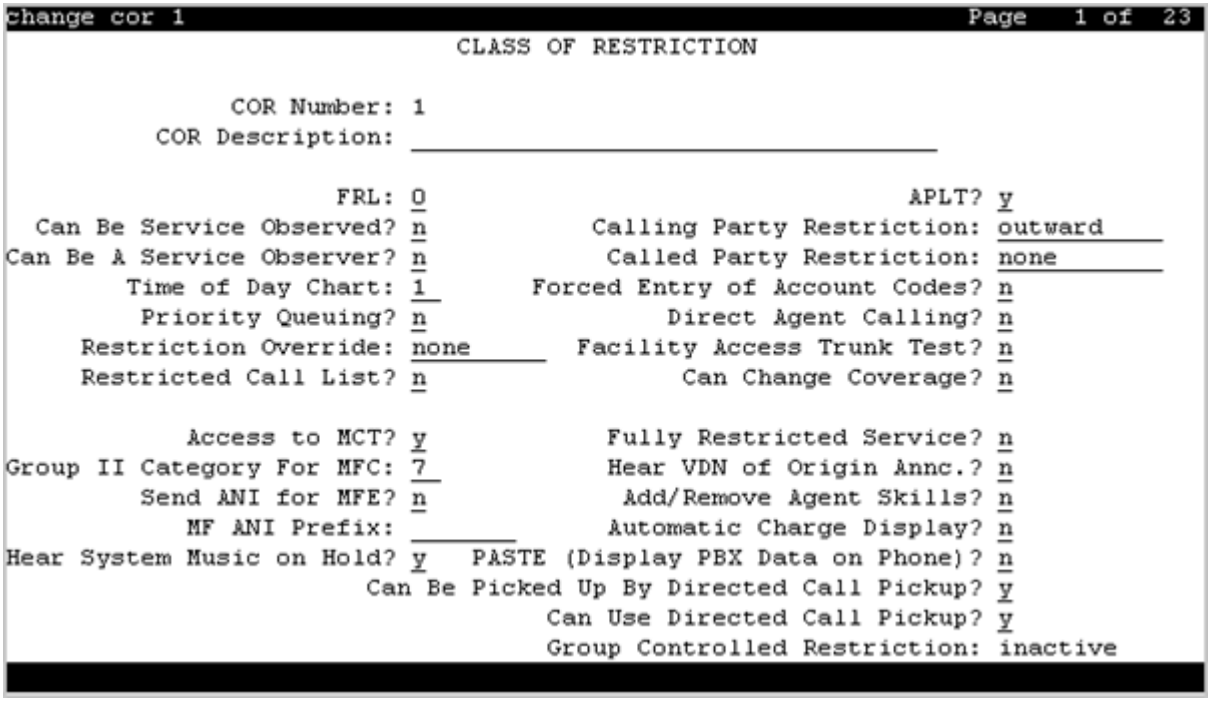
For more information, see *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431.

Chapter 26: Class of Restriction

Establishes classes of restriction (COR). Classes of restriction restrict users from originating or terminating certain types of calls. A system might use only one COR or as many as necessary to control calling privileges.

Example command: `change cor n`, where *n* is the COR number.

Class of restriction: page 1



Access to MCT

Grants or denies access to the Malicious Call Trace feature.

Valid Entry	Usage
y	With this, user can activate a request to trace a malicious call.
n	With this, user cannot request a malicious call trace. If this extension is the subject of a malicious call trace, the MCT History report does not display the extension.

Add/Remove Agent Skills

User can add or remove skills.

APLT

Users can gain access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Unavailable in the U.S.

Valid Entry	Usage
y	Call charges display during and at the end of the call.
n	Call charges display only when users press the **disp-chrg** button before the call drops.

Called Party Restriction

Valid Entry	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls can go through the called party and attendant-assisted calls if administered for this COR.
Termination	Blocks the called party from receiving any calls at any time.
none	No called party restrictions.

Related Links

Restriction Override on page 123

Calling Party Restriction

Determines the level of calling restriction associated with this COR.

Important:

Limit calling permissions as much as possible.

Valid Entry	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A telephone with this COR can initiate Remote Access calls, if the COR of the barrier code supports it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same server running Communication Manager or within a private network. To enhance security, you can use outward restrictions when practical.
All-toll	Blocks the calling party from making ARS and trunk access calls to certain toll areas as defined by the Dialed String on the ARS Toll Analysis table. The call completes if the facility COR is also associated with an Unrestricted Call List and whose Dialed String also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls to certain toll areas as defined by the Dialed String on the ARS Toll Analysis table. The call completes if the facility COR is also associated with an Unrestricted Call List and whose Dialed String also matches the dialed number.
none	No calling party restrictions.

Related Links

Dialed String on page 19

Can Be a Service Observer

A user with this CO permission can be a service observer.

Available only if Basic Service Observing is enabled for the system.

Caution:

Service Observing might be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

Related Links

Service Observing (Basic) on page 810

Can Be Picked Up By Directed Call Pickup

Calls for this station or EAS agent can be picked up using the Directed Call Pickup Up feature.

Available only if Directed Call Pickup is enabled for the system.

Related Links

Directed Call Pickup on page 342

Can Be Service Observed

Service observing can be activated for physical extensions and logical agent IDs and VDNs.

Can Change Coverage

Station users can: Or Station users can have permission to:

- Select one of two previously-administered coverage paths
- Activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location

Can Use Directed Call Pickup

The station, attendant, or EAS agent permission can pick up calls using the Directed Call Pickup feature.

Available only if Directed Call Pickup is enabled for the system.

Related Links

Directed Call Pickup on page 342

COR Description

The description of the COR. Accepts up to 35 characters.

Tip:

A clear description makes it easier to remember which COR to assign when adding users.

Example

Two examples of COR descriptions include "Customer Service" and "Legal Department".

COR Number

The COR number.

Direct Agent Calling

Users can use the Direct Agent Calling feature. Using Direct Agent Calling, users can dial an ACD agent extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent can receive calls directly.

Facility Access Trunk Test

Users can perform Facility Access Trunk Tests. When this feature is active, the **trk-ac-alm** feature button status lamp lights when a successful test attempt occurs.

Forced Entry of Account Codes

If enabled, the Forced Entry of Account Codes (FEAC) feature is active on this COR. Any telephone assigned the associated COR must dial an account code before making an outgoing call. If a call is being routed by ARS, the system prevents account code checking on the COR.

Available only when the FEAC feature is enabled for the system.

Important:

If a COR using the FEAC feature is assigned to a VDN, the route-to commands executed by the associated vector will be unsuccessful.

Related Links

Forced Entry of Account Codes on page 800
CDR FEAC on page 853

FRL

Security alert:

Assign the lowest possible FRL to enhance system security.

Valid Entry	Usage
0 to 7	Assigns a Facilities Restriction Level (FRL) to the COR to enhance system security. AAR or ARS features use this entry to determine call access to an outgoing trunk

Valid Entry	Usage
	group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator. The call originator is typically a telephone user. An originating FRL of 0 has the least calling privileges.

Fully Restricted Service

Unavailable for Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities.

Valid Entry	Usage
y	Stations do *not* have access to the public network for either incoming or outgoing calls.
n	Stations have access to the public network for either incoming or outgoing calls.

Group II Category For MFC

Valid Entry	Usage
1 to 10	Controls categories for Russian signaling trunks. Also controls categories for R2-MFC signaling trunks if administered system-wide. The server running Communication Manager sends this value as the Calling or Called Party Category for telephones or trunks that use this COR. This administered Calling Party Category digit is included as part of the ANI information sent to the local telephone company central office on request using R2-MFC signaling.

Related Links

Use COR for Calling Party Category on page 578

Group Controlled Restriction

Valid Entry	Usage
active	The current COR is under controlled restriction.
inactive	The current COR is beyond controlled restriction.

Hear System Music on Hold

If you enable the **Music on Hold** feature on the telephone, callers that the user puts on hold will hear music.

Hear VDN of Origin Announcement

Agents can receive VDN of Origin Announcement (VOA) messages that provide agents with a short message about the caller's city of origin or requested service based on the VDN used to process the call.

MF ANI Prefix

The prefix applied to an extension number when ANI is sent to the local telephone company central office. Accepts up to seven digits. This COR-specific value overrides any ANI prefix administered system-wide for multifrequency signaling. The prefix is inapplicable when ANI is tandemed through the Communication Manager server on tandem calls. Also applies to the ANI for the server when the originating side is a trunk and there was no ANI.

Related Links

ANI Prefix on page 571

Partitioned Group Number

Available only if AAR/ARS Partitioning is enabled for the system and partition groups are administered.

Valid Entry	Usage
1 to 8	The AAR/ARS partitioned group number associated with this COR.

Related Links

ARS/AAR Partitioning on page 794

PASTE (Display PBX Data on telephone)

Use this field to enable the PASTE feature.

Valid entry	Usage
y	Downloads all lists.
n	Disables the PASTE feature.

Priority Queuing

If enabled, a telephone user’s calls are placed ahead of non-priority calls in a hunt group queue.

Related Links

ACD on page 807

Restricted Call List

Use this field to allow or restrict access to the Restricted call list.

Valid entry	Usage
y	Users cannot make calls to numbers on the Restricted call list from a facility assigned to this COR.
n	Users can make calls to numbers on the Restricted call list from a facility assigned to this COR.

Related Links

RCL on page 855

Restriction Override

Users assigned to this COR can override inward restriction during conference, transfer or call forwarding operation to a telephone that is inward restricted.

Valid Entry	Usage
attendant	Attendants might override inward restrictions. A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls.
all	All users can override inward restrictions
none	No users can override inward restrictions

Send ANI for MFE

Applicable only:

- For Spain
- For 2/6 signaling, but not 2/5 signaling

Valid Entry	Usage
y	Enables Automatic Number Identification (ANI) to send the calling party's number to the public or IBERCOM network so that charges are broken down by line.
n	Charges are not itemized by line. The company receives a single bill for the total number of calls made; also called *block charging*.

Time of Day Chart

Available only if Time of Day Routing is enabled for the system.

Valid Entry	Usage
1 to 8	The AAR/ARS time-of-day-chart number associated with this COR.

Related Links

Time of Day Routing on page 806

Unrestricted Call List

Valid Entry	Usage
1 to 10 blank	Overrides specified toll call restrictions for a COR otherwise restricted from making ARS or trunk access calls.

Related Links

Dialed String on page 19

Class of Restriction: page 2

```
change cor 1                                                Page   2 of  23
                              CLASS OF RESTRICTION

                      MF Incoming Call Trace? n
                Brazil Collect Call Blocking? n
                      Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                      Remote Logout of Agent? n


 Station Lock COR: 1___      TODSL Release Interval (hours):
                                    ASAI Uses Station Lock? n
        Line Load Control: 1
 Maximum Precedence Level: ro           Preemptable? y
 MLPP Service Domain: ________

      Station-Button Display of UUI IE Data? n
      Service Observing by Recording Device? n
              Can Force A Work State Change? n
            Work State Change Can Be Forced? n
               Restrict Second Call Consult? n
```

ASAI Uses Station Lock

Valid Entry	Usage
y	Enables the ASAI originated calls to follow the restriction or permission assigned to the locked station, via the Station Lock COR feature.
n	Enables the ASAI originated calls to follow the permission or restrictions of the COR assigned to the endpoint. The default value is n.

Block Enhanced Conference/Transfer Display

Valid entry	Usage
y	Blocks all the enhanced conference and transfer display messages on digital telephones, except the Transfer Completed message.
n	Does not block the enhanced conference and transfer display messages.

Block Transfer Displays

Valid Entry	Usage
y	Users of DCP, Hybrid, ISDN-BRI, or wireless display telephones cannot receive a confirmation message when they transfer a call.
n	Users can receive a confirmation message when they transfer a call.

Brazil Collect Call Blocking

If enabled, all Brazilian trunk calls that terminate to a station send back a double answer to the local telephone company central office (CO). This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call.

Line Load Control

Valid Entry	Usage
1 to 4	Specifies the line load control level for this COR, where 1 has no restrictions and 4 is the most restrictive.

Maximum Precedence Level

Assigns a maximum precedence level for extensions with this COR for use with the Multiple Level Precedence and Preemption feature.

Valid Entry	Usage
fo	Flash Override
fl	Flash
im	Immediate
pr	Priority
ro	Routine. This is the default value.

MF Incoming Call Trace

If enabled, a Call Trace can be assigned to a station. Communication Manager then generates an MFC backward signal during call setup instead of the “free” signal. This triggers the local telephone company central office to collect trace information before releasing the calling party.

MLPP Service Domain

Valid Entry	Usage
1 to 16777215	The service domain for users and trunks to which this COR is assigned

Outgoing Trunk Alerting Timer (minutes)

Applies an alerting tone to an outgoing trunk call after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before an alerting tone is applied to the call.
blank	The Alerting Tone for Outgoing Trunk Calls feature is disabled and the alerting tone is not applied to the call. This is the default.

Related Links

Trunk Alerting Tone Interval (seconds) on page 294

Outgoing Trunk Disconnect Timer (minutes)

Disconnects an outgoing trunk automatically after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before automatically disconnecting the call. A warning tone is given to all parties on the trunk call 1 minute before the administered value and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.
blank	Outgoing trunk calls disconnect only when dropped by one or all parties. This is the default.

Preemptable

If enabled, makes extensions with this COR preemptable for Multiple Level Precedence and Preemption calls.

Remote Logout of Agent

If enabled, users can use a feature access code to logout an idle ACD or EAS agent without being at the agent's telephone.

Restrict Second Call Consult

Use this field to enable the Restrict Second Agent Consult feature. With this feature, Communication Manager allows the agents of Avaya Aura® Contact Center to use only one consult operation at a time. The consult operations include transfer and conference.

Note:

If you set the **Restrict Second Call Consult** field to `n`, after you dial the **Restrict Second Consult** feature access code, the feature access code overrides the setting of the **Restrict Second Call Consult** field on the COR group of the station.

Service Observing by Recording Device

If enabled, the service observer associated with this COR can use an audio recording device, such as the Witness product.

Station-Button Display of UUI IE Data

If enabled, a station user can push a **uui-info station** button and see up to 32 bytes of ASAI-related User-User-Information Information Element (UUI-IE) data. Pressing the **uui-info** button displaces the incoming call/collected digits display. Pressing **callr-info** re-displays the collected digits.

Available only with Call Center release 3.0 or later.

Related Links

Call Center Release on page 808

Station Lock COR

Valid Entry	Usage
0 to 995	If the user dials the station lock Feature Access Code (FAC), the user's extension applies the Station Lock COR for call restrictions instead of the station's usual COR. For information on call restrictions, see the Station Lock section of *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205. This field defaults to the current COR.

Related Links

Station Lock Activation/Deactivation on page 254

Class of Restriction: page 3

The Send All Calls and Call Forwarding (SAC/CF) Override feature overrides active rerouting. This feature overrides these active rerouting settings:

- Send All Calls (SAC)
- Call Forwarding (CF) all
- Enhanced Call Forwarding (ECF) unconditional

```
change cor 1                                                  Page   3 of  23
                              CLASS OF RESTRICTION

                          ERASE 24XX/14XX/94XX USER DATA UPON
            Dissociate or unmerge this phone: none
           EMU login or logoff at this phone: none
            Mask CPN/NAME for Internal Calls? n

               SAC/CF Override by   Team Btn? n  Priority Call? n  Dialing? n
   SAC/CF Override Protection for   Team Btn? n  Priority Call? n  Dialing? n



                         one-X Server Access? y
Team Btn  Silent if Active? n  Priority Ring? n
Team Btn  Display Name? n      Pick Up by Going Off Hook? n

   Outgoing Trunk Disconnect Timer (minutes):
     Outgoing Trunk Alerting Timer (minutes):
```

Auto Answer

Enables or disables automatic answer on team button calls.

Erase 24xx/14xx/94xx User Data Upon: Dissociate or unmerge this phone

Administers what local terminal data items are erased when the 24*xx*, 14*xx*, or 94*xx* is dissociated or unmerged.

Valid Entry	Usage
none	Default value. No local terminal data is erased.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Erase 24xx/14xx/94xx User Data Upon: EMU login or logoff at this phone

Administers what local terminal data items are erased upon Enterprise Mobility User (EMU) login or logoff.

Valid Entry	Usage
none	This is the default. No local terminal data is erased.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Mask CPN/Name for Internal Calls

Valid entry	Usage
y	Hides the display of calling and called party numbers and administered name on internal calls.
n	Shows the display of calling and called party numbers and administered name on internal calls.

one-X Server Access

Enables or disables the one-X Server Access for one-X control.

Outgoing Trunk Disconnect Timer (minutes)

Disconnects an outgoing trunk automatically after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before automatically disconnecting the call. A warning tone is given to all parties on the trunk call 1 minute before the administered value and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.
blank	Outgoing trunk calls disconnect only when dropped by one or all parties. This is the default.

Priority Ring

Enables or disables priority ringing for speed dialing on team button calls.

SAC/CF Override by Priority Call and Dialing

Use this field to enable or disable the SAC/CF override by Priority Call and Dialing feature.

SAC/CF Override by Team Btn

Use this field to allow the user of a monitoring station to override any active rerouting, such as Send All Calls or Call Forwarding, on the monitored station.

The monitoring station can override the rerouting only if the following conditions are met:

- The **SAC/CF Override by Team Btn** field is set to y.
- The **SAC/CF Override Protection for Team Btn** field is set to n.
- The monitoring station uses the **team** button the make the call to the monitored station.

Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

SAC/CF Override Protection for Priority Call and Dialing

Use this field to enable or disable the SAC/CF Override Protection for Priority Call and Dialing feature.

SAC/CF Override Protection for Team Btn

Use this field to restrict the user of a monitoring station from overriding the active rerouting, such as Send All Calls or Call Forwarding, on the monitored station. This field works only when the monitoring station uses the **team** button the make a call to the monitored station.

Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

Team Btn Display Name

Enables or disables the display of station name on team button calls.

Team Btn Silent if Active

Enables or disables audible ringing on team button calls.

Team Pick Up by Going Off Hook

Enables or disables pick-up by going off hook on team button calls.

Class of restriction: page 4 to page 23

You can use the screens from page 4 to page 13 to assign calling permissions and the screens from page 14 to page 23 to assign service observing permissions to the COR that you administer.

CALLING PERMISSION

```
change cor 1                                                  Page   4 of  23
                                CLASS OF RESTRICTION

CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

  0? y      15? y      30? y      44? y      58? y      72? y      86? y
  1? y      16? y      31? y      45? y      59? y      73? y      87? y
  2? y      17? y      32? y      46? y      60? y      74? y      88? y
  3? y      18? y      33? y      47? y      61? y      75? y      89? y
  4? y      19? y      34? y      48? y      62? y      76? y      90? y
  5? y      20? y      35? y      49? y      63? y      77? y      91? y
  6? y      21? y      36? y      50? y      64? y      78? y      92? y
  7? y      22? y      37? y      51? y      65? y      79? y      93? y
  8? y      23? y      38? y      52? y      66? y      80? y      94? y
  9? y      24? y      39? y      53? y      67? y      81? y      95? y
 10? y      25? y      40? y      54? y      68? y      82? y      96? y
 11? y      26? y      41? y      55? y      69? y      83? y      97? y
 12? y      27? y      42? y      56? y      70? y      84? y      98? y
 13? y      28? y      43? y      57? y      71? y      85? y      99? y
 14? y      29? y
```

Users with this COR can call users with the CORs from 0 to 995.

Valid entry	Usage
y	A station user with the COR that you administer can call stations with the other COR. The default value is y.
n	A station user with the COR that you administer cannot call stations with the other COR.

SERVICE OBSERVING PERMISSION

```
change cor 1                                                  Page  14 of  23
                              CLASS OF RESTRICTION

SERVICE OBSERVING PERMISSIONS
(Enter "y" to grant permission to service observe the specified COR)

  0? y      15? y      30? y      44? y      58? y      72? y      86? y
  1? y      16? y      31? y      45? y      59? y      73? y      87? y
  2? y      17? y      32? y      46? y      60? y      74? y      88? y
  3? y      18? y      33? y      47? y      61? y      75? y      89? y
  4? y      19? y      34? y      48? y      62? y      76? y      90? y
  5? y      20? y      35? y      49? y      63? y      77? y      91? y
  6? y      21? y      36? y      50? y      64? y      78? y      92? y
  7? y      22? y      37? y      51? y      65? y      79? y      93? y
  8? y      23? y      38? y      52? y      66? y      80? y      94? y
  9? y      24? y      39? y      53? y      67? y      81? y      95? y
 10? y      25? y      40? y      54? y      68? y      82? y      96? y
 11? y      26? y      41? y      55? y      69? y      83? y      97? y
 12? y      27? y      42? y      56? y      70? y      84? y      98? y
 13? y      28? y      43? y      57? y      71? y      85? y      99? y
 14? y      29? y
```

Users with this COR can observe users with the CORs from 0 to 995.

Valid entry	Usage
y	A station user with the COR that you administer can observe stations with the other COR. The default value is y.
n	A station user with the COR that you administer cannot observe stations with the other COR.

Chapter 27: Class of service

Administers access permissions for call processing features that require dial code or feature button access.

Note:

Class of Service (COS) is inapplicable to trunk groups except for the Remote Access feature.

A COS assignment defines which features and functions a telephone user can access. Up to 16 different COS numbers can be administered (0 to 15). When Tenant Partitioning is enabled for the system, you can administer up to 100 COS groups, each with 16 Classes of Service.

The screen lists the default values for each COS or feature combination. For a particular combination, y is for access to the feature and n is for denying access.

Caution:

Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Example command: `change cos-group` *n*, where *n* is the COS group number.

Related Links

Tenant Partitioning on page 806

Class of service: page 1

```
change cos-group 1                                          Page   1 of   2
CLASS OF SERVICE          COS Group: 1    COS Name: ________________

                          0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
Auto Callback             n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
Call Fwd-All Calls        n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
Data Privacy              n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
Priority Calling          n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
Console Permissions       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Off-hook Alert            n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Client Room               n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Restrict Call Fwd-Off Net y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
Call Forwarding Busy/DA   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Personal Station Access (PSA) n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Extended Forwarding All   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Extended Forwarding B/DA  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Trk-to-Trk Transfer Override n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
QSIG Call Offer Originations n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Contact Closure Activation n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

Automatic Callback

Enables or disables Automatic Callback. Using Automatic Callback, internal users who place a call to a busy or an unanswered internal telephone can be automatically called back when the called telephone becomes available.

Automatic Exclusion

Available when Automatic Exclusion is enabled for the system and the applicable station has an assigned **Exclusion** button.

Valid Entry	Usage
y	Activates Automatic Exclusion when a user goes off hook on a station. Automatically prevents other multi-appearance users from bridging onto a call.
n	Denies Automatic Exclusion. Manual exclusion is available when the user presses the **Exclusion** button before dialing or during a call.

Related Links

[Automatic Exclusion by COS](#) on page 333

Call Forwarding All Calls

Used to activate the call forwarding feature for all calls to any extension.

Call Forwarding Busy/DA

If enabled, users can forward calls to any extension when the dialed extension is busy or does not answer.

Client Room

Users can access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. Available only with Hospitality Services and a Property Management System interface.

Console Permissions

Multi-appearance telephone users can control the same features that the attendant controls.

Contact Closure Activation

A user can open and close a contact closure relay. Contact closures control electrical devices remotely. Users dial a Feature Access Code (FAC) on a telephone to activate electrical devices such as electrical door locks.

COS Group

To administer a COS group, Tenant Partitioning must be enabled for the system.

Valid Entry	Usage
1 to 100	The COS Group number.

Related Links

Tenant Partitioning on page 806

COS Name

The identifying name for this COS group. To administer a COS group, Tenant Partitioning must be enabled for the system.

Related Links

Tenant Partitioning on page 806

Data Privacy

If activated, a user can enter a Feature Access Code (FAC) that protects a data call from being interrupted by any of the system override or ringing features.

Extended Forwarding All

A user can administer call forwarding from a remote location for all calls. Available only if Extended Coverage and Forwarding Administration is enabled for the system.

Related Links

Extended Cvg/Fwd Admin on page 799

Extended Forwarding B/DA

A user can administer call forwarding from a remote location when the dialed extension is busy or does not answer. Available only if Extended Coverage and Forwarding Administration is enabled for the system.

Related Links

Extended Cvg/Fwd Admin on page 799

Off-Hook Alert

Enables or disables the requirement that the system send an emergency call to the attendant if the telephone remains off-hook for a prescribed time. Available only if Basic Hospitality or Emergency Access to Attendant is enabled for the system.

Related Links

Hospitality (Basic) on page 800
Emergency Access to Attendant on page 798

Personal Station Access (PSA)

Users can associate a telephone with their extension using their programmed services using a Feature Access Code (FAC). Deactivate for virtual telephones. Required for a user's home station in order for that user to use the Enterprise Mobility User (EMU) feature at other stations. Available only if Personal Station Access (PSA) is enabled for the system.

Related Links

Personal Station Access (PSA) on page 804

Priority Calling

If activated, a user can dial a Feature Access Code (FAC) to originate a priority call. Priority calls ring differently and override send all calls.

QSIG Call Offer Originations

If activated, a user can invoke QSIG Call Offer services.

Restrict Call Fwd-Off Net

Valid Entry	Usage
y	A user cannot forward calls to the public network. **Security alert:** For security reasons, this should be enabled for all classes of service except the ones used for special circumstances.
n	A user can forward calls to the public network.

Trk-to-Trk Restriction Override

Use this field to allow or deny user permission to override the system or COR-to-COR calling party restrictions that prohibit the trunk-to-trunk transfer operation for users with this COS.

Security alert:

Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

Class of service: page 2

```
change cos-group 1                                          Page   2 of   2
                              CLASS OF SERVICE

                              0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
VIP Caller                    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n

Masking CPN/Name Override     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Call Forwarding Enhanced      y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
Priority Ip Video             n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Ad-hoc Video Conferencing     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
MOC Control:                  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Match BCA Display To Principal n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

Ad hoc Video Conferencing

Enables or disables Ad-hoc Video Conferencing, so that up to six users can participate in a video conference call.

Call Forwarding Enhanced

If activated, users can designate different preferred destinations for forwarding calls that originate from internal and external callers.

DCC Activation/Deactivation

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Valid Entry	Usage
y	Activate the Destination Code Control feature.
n	Deactivate the Destination Code Control feature. The default value is `n`.

Masking CPN/Name Override

Users can be assigned or denied users permission to override the MCSNIC capability. MCSNIC masks the display of calling party information and replaces it with a hard-coded system-wide text string, **Info Restricted**.

Match BCA Display to Principal

Valid entry	Usage
y	The display format of incoming calls on the bridged call appearances of a COS group is <calling name/number>.
n	The display format of incoming calls on the bridged call appearances of a COS group is <calling name/number> to <principal station>.

MOC Control

The **MOC Control** field indicates that the station is under Microsoft Office Communicator (MOC) control.

If a station uses the MOC client and also wants to use an office telephone and an off-PBX telephone, such as a mobile phone simultaneously, set the **MOC Control** field to y.

Valid entry	Usage
y	Station is under MOC control.
n	Station is beyond MOC control. This is the default value.

Priority Ip Video

Activate or deactivate priority video calling. Video calls have an increased likelihood of receiving bandwidth and can also be allocated a larger maximum bandwidth per call.

VIP Caller

Enables or disables automatic priority calling when assigned to the originator of a call. A call from a VIP telephone is always a priority call without the use of a feature button or FAC.

Chapter 28: Code Calling IDs

On systems with chime paging, assigns a unique series of chimes to extensions using a chime code. The chime code assigned to an extension plays over the speakers whenever that extension is paged. Assigns chime codes to up to 125 extensions.

Example command: `change paging code-calling-ids`

Ext

Assigns extensions to chime codes. Only one extension can be assigned to each chime code. The extension cannot be assigned to a code that is a Vector Directory Number (VDN).

Chapter 29: Configuration Set

Use the Configuration Set screen to define call treatment options for Extension to Cellular calls. With the Extension to Cellular feature, you can use up to 99 configuration sets that are already defined in the system with default values.

Note:

In the One-X mobile environment, you can edit the values of only the **Cellular voice mail detection** field and the **Call log notify** field. All other fields are read-only.

To modify the values of the fields on the Configuration Set screen, type the following command:

`change off-pbx-telephone configuration-set` *n* , where *n* is the Configuration Set number.

Apply Ringback upon Receipt of

Use the Apply ringback Upon Receipt of field to configure when Communication Manager must send the ringback tone to the caller.

Valid entry	Usage
Call-Proceeding	Communication Manager sends the ringback tone to the caller after receiving the Call-Proceeding message from the EC500 network. Call-Proceeding is the default value.
Alert	Communication Manager sends the ringback tone to the caller after receiving the Alert message from the EC500 network.

Barge-In Tone

Enables or disables a barge-in tone used to add security to Extension to Cellular calls. If a user is on an active Extension to Cellular call and another person joins the call from an Extension to Cellular enabled office telephone, all parties on the call hear the barge-in tone.

Call Appearance Selection for Origination

Specifies how the system selects a Call Appearance for call origination. To use this feature, bridged calls must be enabled for the system.

Valid Entry	Usage
first-available	The system searches for the first available regular or bridged Call Appearance.
primary-first	The system first searches for a regular Call Appearance for call origination. • If all Call Appearances are busy at the primary station, the call cannot be initiated. • If the primary station is logged-off, a second search is made that includes both regular and bridged Call Appearances. This is the default.

Related Links

Bridged Calls on page 757

Call Log Notify

Use this field to update the Avaya one-X® Client Enablement Services logs with actual outpulsed digits and not the digits that the user dials.

Valid entry	Usage
dialed-digits	When you enter dialed-digits, the endpoint displays the dialed digits.
outpulsed-digits	When you enter outpulsed-digits, Communication Manager analyzes the dialed digits and sends the digits to Client Enablement Services for call logging.

Calling Number Style

Determines the format of the caller ID for calls from a local Avaya Communication Manager extension to an Extension to Cellular telephone.

Valid Entry	Usage
network	Provides a display of only 10-digit numbers. For internal calls, the ISDN numbering tables are used to create the calling number and DCS calls use the ISDN calling number if provided. The externally provided calling number is used when available for externally originated calls.
pbx	Provides a display of less than 10-digits. Extensions sent as the calling number for all internally- and DCS network-originated calls.

Calling Number Verification

Use this field to impose restrictions on calls originating from an Extension to Cellular-enabled cell phone to some other extension.

Valid entry	Usage
y	If you set the field to y and the network service provider has not marked the screening indicator as network provided or user provided verified and passed, Communication Manager treats the incoming call as an ordinary incoming trunk call and not as a call from an extension.
n	If you set the field to n, Communication Manager treats the incoming call as a call from the corresponding desk phone.

CDR for Calls to EC500 Destination

Determines whether a Call Detail Record (CDR) is generated for any call to the cell telephone. Available only if CDR reports is enabled for the trunk group.

Valid Entry	Usage
y	Treats calls to the XMOBILE station as trunk calls and generates a CDR.
n	Treats calls to the XMOBILE station as internal calls, without generating a CDR.

Related Links

CDR Reports on page 480

CDR for Origination

Determines the CDR report format when CDR records are generated for a call that originates from an Extension to Cellular cell phone. To generate this CDR, you must enable the Incoming Trunk CDR. The CDR report excludes dialed Feature Name Extensions (FNEs).

Valid Entry	Usage
phone-number	The calling party on the CDR report is the 10-digit cell phone number. This is the default.
extension	The calling party on the CDR report is the internal office telephone extension associated with the Extension to Cellular cell phone
none	The system does not generate an originating CDR report.

Cellular Voice Mail Detection

You can use this field if you do not want an Extension to Cellular call to cover to the cellular voice mail. When the call server detects that the call has covered to the cellular voice mail, the call server brings the call back to the server.

Valid entry	Usage
None	The call covers to the cellular voice mail. This entry is the default value.
Timed	The system displays the **Seconds** field. You can enter any value from 1 to 9. If the user answers the call on the cellular phone within the configured number of seconds, Communication Manager disconnects the cellular leg of the call. The call continues to ring at the desk phone. You can use this configuration for GSM, CDMA, and ISDN networks.
Message	The message option works with carriers who use non-ISDN voice mail systems. You must not use this option with ISDN-based voice mail systems.

Configuration Set Description

A description of the purpose of the configuration set. Accepts up to 20 alphanumeric characters.

Example

“Extension to Cellular handsets”

Confirmed Answer

Enables or disables Confirmed Answer on Extension to Cellular calls for this station. If enabled, requires the user to input a digit to confirm receipt of a call sent to a cellular telephone by the Extension to Cellular feature. Upon answering the incoming call on the cellular telephone, the user hears a dial tone. The user must then press any one of the digits on the telephone keypad. Until the system receives a digit, the system treats the call as unanswered. The length of time to wait for the digit can be administered from 5 to 20 seconds, with a default of 10 seconds. The system plays a recall dial-tone to indicate that input is expected. During the response interval, the original call continues to alert at the desk set and any stations bridged to the call. If the user does not enter a digit before the time-out interval expires, the call is pulled back from the cell telephone.

Fast Connect on Origination

Determines whether additional processing occurs on Communication Manager prior to connecting a call. Reserved for future capabilities provided by the cell telephone provider but currently not used.

Location for routing incoming overlap calls

With the Location for routing incoming overlap calls feature, Communication Manager allows the administrator to configure the location to route incoming calls that are mapped to EC500, CSP, and Avaya one-X applications over ISDN-PRI, ISDN-BRI, and H.323 overlap trunks.

The overlap trunk refers to an ISDN trunk group configured for overlap receiving. The **Digit Handling (in/out)** field on the ISDN Trunk Group screen is set to `overlap/overlap` or `overlap/enbloc`.

Valid entry	Usage
Station	Communication Manager uses the station location to process all incoming calls that are initiated from an EC500-mapped phone. Communication Manager uses the station location only if the location is configured on the EC500-mapped station screen. `Station` is the default value.
Trunk	Communication Manager uses the incoming trunk location to process all incoming calls that are initiated from an EC500-mapped phone.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are mapped into XMOBILE stations. These options come into effect after the call has entered the active state (Communication Manager has sent a CONNECT message back to the network).

Valid Entry	Usage
dtmf	Expect digits from either in-band or out-of-band, but not simultaneously. The server allocates a DTMF receiver whenever it needs to collect digits. This option normally would be used for Extension to Cellular XMOBILE station calls.
out-of-band	Expect all digits delivered by out-of-band signaling only. Communication Manager collects digits that it needs from the out-of-band channel (no touch-tone receiver). In addition, any digits received when the server is not collecting digits are converted to DTMF and broadcast to all parties on the call. This option is in force for DECT XMOBILE station calls.

Table continues…

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Valid Entry	Usage
both	Expect all subsequent digits delivered by simultaneous in-band and out-of-band signaling. Out-of-band signaling consists of digits embedded in ISDN INFO messages while the in-band signaling consists of DTMF in the voice path. Communication Manager collects all digits that it needs from the out-of-band channel. No touch tone receive is allocated to prevent collecting double digits. End-to-end signaling occurs transparently to the server via in-band transmission of DTMF. This option is in force for PHS XMOBILE station calls.

Provide Forced Local Ringback for EC500

When a caller calls an enterprise user, the caller expects a standard ringback tone. If the EC500 feature is activated on the telephone of the enterprise user and if the user uses custom ringback tones on the cell phone, the caller hears the custom ringback tone. If you set the **Provide Forced Local Ringback for EC500** field to y, Communication Manager provides the local ringback tone and ignores the custom ringback tone of the cell phone.

Chapter 30: Console parameters

Administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays.

Example command: `change console-parameters`

Console parameters: page 1

AAR/ARS Access Code

An optional AAR/ARS access code used to route to the main switch. Accepts an up to four-digit access code. Available only for a QSIG branch CAS configuration.

Related Links

CAS on page 150

Allow Night Service Calls To Forward

Enables or disables forwarding of DID calls to station users. When Station A receives a DID call and Station A does not answer the call, DID times out and the system routes the call to an attendant. When the attendant is in Night Service mode, the system routes the call to a night service station. If Call Forward is active on the night service station, the system routes the call to Station B. When Call Forward is inactive, the night service station will continue to ring.

Valid Entry	Usage
y	Enables forwarding of DID calls from the night service station.
n	Disables the forwarding of DID calls to station users. This is the default value.

Alternate FRL Station

The extension of the alternate facility restriction level (FRL) activation station.

Attendant Group Name

A name for the attendant group. Accepts up to 27 alphanumeric characters.

Attendant Lockout

Activates or deactivates Privacy — Attendant Lockout. Privacy — Attendant Lockout prohibits an attendant from reentering a conference call that has been placed on hold unless recalled by a telephone user on the call. This feature provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Attendant Vectoring VDN

Assigns the VDN extension for Attendant Vectoring to a console. Available only if Attendant Vectoring is enabled for the system and Tenant Partitioning is disabled.

Related Links

Attendant Vectoring on page 796
Tenant Partitioning on page 806

Backup Alerting

Used to assign or remove permissions for system users to pick up alerting calls if the attendant queue has reached its warning state.

Calls In Queue Warning

The number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The queue maximum depends on system capacities.

CAS

Using the Centralized Attendant Service (CAS), users at separate locations can concentrate attendant positions at one location. Incoming trunk calls to unattended branch locations are routed to the main attendant.

Valid Entry	Usage
main	The main Communication Manager server where the attendant group is located. Uses non-ISDN signaling. CAS Main must be enabled for the system.
branch	A branch Communication Manager server. There are no local attendants, so attendant-seeking calls route to the main Communication Manager server. Uses non-ISDN signaling. CAS Branch must be enabled for the system
none	Disables CAS.
QSIG-main	Same as main, but with QSIG signaling among the Communication Manager servers. Centralized Attendant must be enabled for the system.
QSIG-branch	Same as branch, but with QSIG signaling among the Communication Manager servers. Centralized Attendant must be enabled for the system.

Related Links

Centralized Attendant on page 814
RLT Trunk Group No. on page 154

CAS Back-Up Ext

Valid Entry	Usage
Extension number Individual attendant console Hunt group TEG	Extension in the dial plan that handles attendant-seeking calls if the RLT trunk group to the CAS Main server is out of service or if CAS Back-Up is activated. Neither a prefixed extension nor a VDN extension is supported. Available only for the CAS Branch feature.

Related Links

CAS Branch on page 796
CAS Main on page 796
Centralized Attendant on page 814

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number for all attendant consoles. The COR for the individual Attendant Console overrides this value.

COS

Valid Entry	Usage
1 to 15	The class of service (COS) number that reflects the required features for all your attendant consoles. The COS for the individual Attendant Console overrides this value.

DID-LDN Only to LDN Night Ext

Valid Entry	Usage
y	Only listed directory number (LDN) calls go to the listed directory night service extension.
n	All attendant seeking calls route to the LDN night service extension.

Ext Alert Port (TAAS)

The seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

An x indicates that there is no hardware associated with this port assignment. If an x is used, there should also be an value for **Ext Alert (TAAS) Extension**.

Related Links

Ext Alert (TAAS) Extension on page 152

Ext Alert (TAAS) Extension

This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned either through TTI or by entering the Ext Alert Port, the extension is automatically removed and treated as unassigned. Available only when the Ext Alert Port (TAAS) has associated hardware.

Related Links

Ext Alert Port (TAAS) on page 152

IAS Att. Access Code

The extension number of the attendant group at the main server running Communication Manager. Required when IAS Branch is enabled. Unavailable if the Centralized Attendant feature is enabled.

Related Links

IAS (Branch) on page 153
Centralized Attendant on page 814

IAS (Branch)

Enables or disables the Inter-PBX Attendant Service (IAS) Branch feature. Unavailable if Centralized Attendant is enabled.

Related Links

Centralized Attendant on page 814

IAS Tie Trunk Group No.

Unavailable if the Centralized Attendant feature is enabled.

Valid Entry	Usage
1 to 2000	The number of the tie trunk group to the main for the IAS (Branch). Required when IAS Branch is enabled.

Related Links

Centralized Attendant on page 814

Night Service Act. Ext.

The extension of the current night service activation station, if any. The station is administered by assigning it a **night-serv** button.

QSIG CAS Number

Contains the complete number of the attendant group at the main server running Avaya Communication Manager, or a Vector Directory Number (VDN) local to the branch server. Accepts up to 20 digits. Cannot be left blank. Available only for an QSIG-branch CAS configuration.

Related Links

CAS on page 150

RLT Trunk Group No.

The trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service. Available only for a branch CAS configuration.

Related Links

CAS on page 150

Console parameters: page 2

ABBREVIATED DIALING

List1, List2, List3

Assigns up to three abbreviated dialing lists to each attendant. A personal list cannot be assigned to an attendant.

Valid Entry	Usage
enhanced	The attendant can access the enhanced system abbreviated dialing list.
group	The attendant can access the specified group abbreviated dialing list. A group number is required.
system	The attendant can access the system abbreviated dialing list.

SAC Notification

Enables or disables Enhanced Attendant Notification for Send All Calls.

COMMON SHARED EXTENSIONS

Busy Indicator for Call Parked on Analog Station Without Hardware?

Enables or disables the Busy Indicator lamp that lights for incoming calls parked on Administration Without Hardware (AWOH) stations.

Count

Valid Entry	Usage
1 to 1182 blank	The number of consecutive extensions, beginning with the Starting Extension used as common, shared extensions

Example

If you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

Related Links

Starting Extension on page 155

Starting Extension

The first extension number in a group of consecutive extensions that can be used by the attendant to park calls.

Related Links

Count on page 154

INCOMING CALL REMINDERS

Alerting (sec)

The number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service.

No Answer Timeout (sec)

Valid Entry	Usage
10 to 1024 blank	The number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. With this option, the call rings for 5 seconds at all points in a coverage path to ensure that the entire path is completed before the call returns to the console.

Secondary Alert on Held Reminder Calls?

Valid Entry	Usage
y	Begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and then switch to secondary alerting when an administered timeout expires.

Related Links

No Answer Timeout (sec) on page 155

TIMING

Overview timer to Group Queue (sec)

Valid Entry	Usage
10 to 1024	The number of seconds a returning call queues to the individual attendant before overflowing to the group. The value applies if the attendant who previously handled the call is busy or unavailable.
blank	The call immediately goes to the group.

Return Call Timeout (sec)

Valid Entry	Usage
10 to 1024 blank	The time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. With this option, the call rings for 5 seconds at all points in a coverage path to ensure that the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Valid Entry	Usage
9 to 999 blank	The number of seconds a call can remain in the attendant queue before activating an alert.

Time Reminder on Hold (sec)

Valid Entry	Usage
10 to 1024	The number of seconds a call can remain on Hold.

Console parameters: page 3

Call-Type Ordering Within Priority Levels?

Groups calls to the attendant in the following order:

1. Queue priority level
2. Call type
3. Order received

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Call type	Description
Type 1 call	Outgoing public-network calls receiving answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
Type 2 call	Incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires.
Type 3 call	All other calls (internal calls, conference calls, and tie-trunk calls of any type). Note that external public-network calls have priority over all other calls including conference calls. Answered public-network calls have priority over unanswered calls.

Valid Entry	Usage
y	Orders calls by call type within each queue priority category. A **type-disp** button is assigned to the Attendant Console. The attendant can review the call type for the active call.
n	Calls are queued in chronological order by queue priority level.

QUEUE PRIORITIES

Assigns a priority level from 1 through 13 to each call category when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

The same priority level can be assigned to more than one call.

Assistance Call

Valid Entry	Usage
1 to 13	Assigns a priority level for assistance calls. A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated. Priority 1 is the highest priority.

DID to Attendant

Valid Entry	Usage
1 to 13	Assigns a priority level for DID to Attendant calls that are incoming DID trunk calls to an attendant group. This excludes trunk calls that return to the attendant group after a timeout or deferred attendant recall. Priority 1 is the highest priority.

Emergency Access

Valid Entry	Usage
1 to 13	Assigns a priority level for emergency access calls that are calls from a telephone user who dials the emergency access code. The default is 1.

Valid Entry	Usage
	Priority 1 is the highest priority.

Individual Attendant Access

Valid Entry	Usage
1 to 13	Assigns a priority level for Individual Attendant Access calls that are calls from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available. Priority 1 is the highest priority.

Interposition

Valid Entry	Usage
1 to 13	Assigns a priority level for Interposition calls that are calls from one attendant to the Individual Attendant Access (IAA) extension of another attendant. Priority 1 is the highest priority.

Miscellaneous Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any unlisted calls. Priority 1 is the highest priority.

Redirected Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls assigned to one attendant, but redirected to the attendant group because the attendant is now busy. Priority 1 is the highest priority.

Redirected DID Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any DID or ACD calls that time out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable. Calls are rerouted to the attendant group. Priority 1 is the highest priority.

Return Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls returned to the attendant after timing out. If the attendant is now busy, the call redirects to the attendant group. Priority 1 is the highest priority.

Serial Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group. Priority 1 is the highest priority.

Tie Call

Valid Entry	Usage
1 to 13	Assigns a priority level for incoming TIE trunk calls (dial-repeating or direct types) to an attendant group. This excludes trunk calls that return to the attendant group after a timeout or deferred attendant recall. Priority 1 is the highest priority.

VIP Wakeup Reminder Call

Valid Entry	Usage
1 to 13	Assigns a priority level for VIP Wakeup Reminder Calls from the Hospitality feature that send a wake-up reminder to the attendant to call a hotel or motel room. Priority 1 is the highest priority.

Console parameters: page 4

QUEUE PRIORITIES

Flash

Valid Entry	Usage
1 to 17	The queue priority for Flash precedence level calls.

Flash Override

Valid Entry	Usage
1 to 17	The queue priority for Flash Override precedence level calls.

Immediate

Valid Entry	Usage
1 to 17	The queue priority for Immediate precedence level calls.

Priority

Valid Entry	Usage
1 to 17	The queue priority for Priority precedence level calls.

Console parameters: page 5

Note:

If MLPP is disabled, the system does not display the MLPP Queues page, but displays page with the following message:

```
Use the 'list attendant' command to see all administered attendants.
```

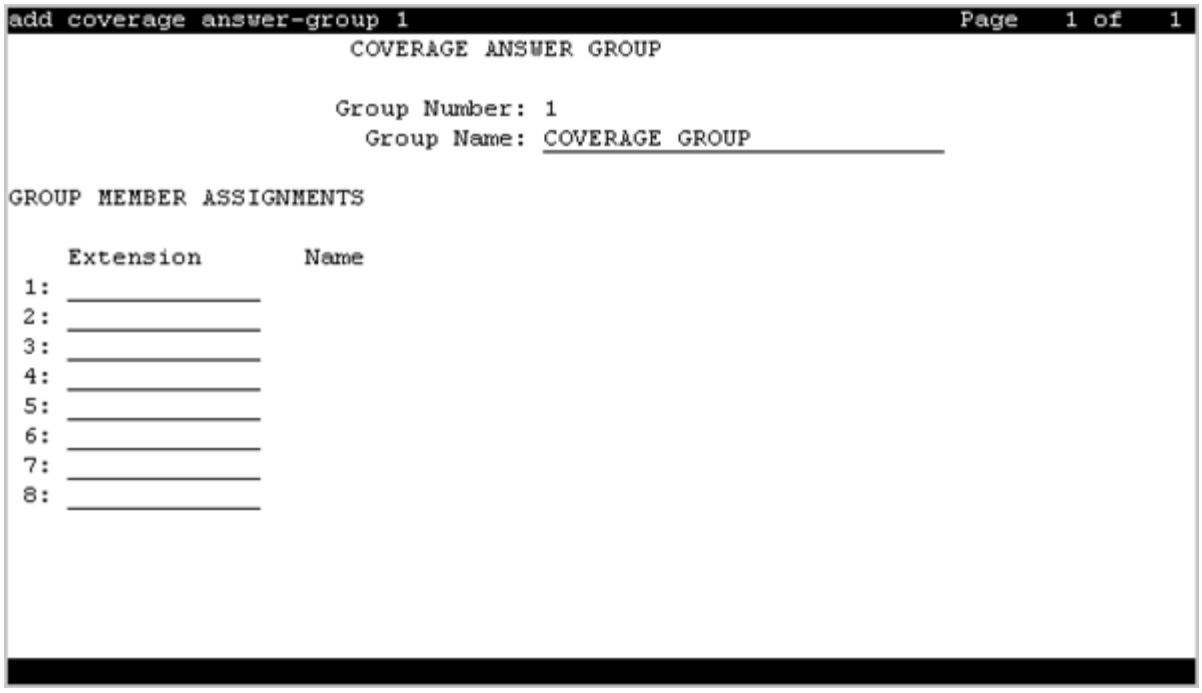
Chapter 31: Coverage Answer Group

Establishes Call Coverage Answer Groups.

An answer group contains up to 100 members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and the department's coverage path displays that group number. All telephones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration. The members of the group are identified by their extension number. Any telephone, including those administered without hardware (but not attendants) can be assigned to a coverage answer group.

Example command: `change coverage answer-group` *n*, where *n* is the assigned group number.



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Ext

The extension number for each member of this coverage answer group. This number cannot be a Vector Directory Number (VDN) extension.

Group Name

The group name used to identify this group. Accepts up to 27 characters. Use the extension numbers of group members as the group name to help when determining which stations are involved in call coverage and trunk coverage paths.

Example

“typing pool”, “room 12”, “secy”

Group Number

The number associated with the Cover Answer Group.

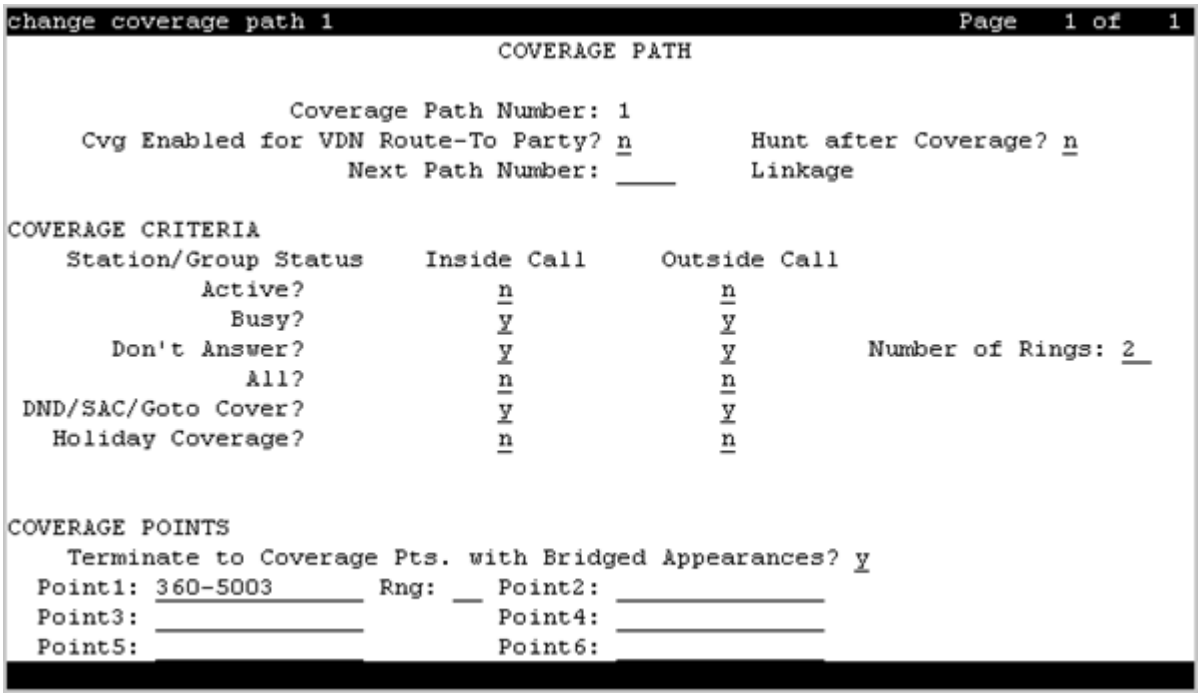
Name

The name assigned when the member’s telephone was administered.

Chapter 32: Coverage Path

Implements Call Coverage Paths by providing the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's telephone rings before the call redirects to coverage.

Example command: `change coverage path n`, where *n* is the assigned coverage path number.



Coverage Path Number

The coverage path being administered.

Cvg Enabled for VDN Route-To Party

Enables or disables the route-to party coverage path after a covered call hits a VDN vector route-to step. By default, the value is n.

Holiday Coverage

Use the **Holiday Coverage** field to redirect all calls during a holiday to a coverage path. For **Holiday Coverage** to function, set the **Don't Answer** field to y.

You must set the **Holiday Coverage** field separately for internal and external calls.

Valid Entry	Usage
y	Communication Manager checks the Holiday Table screen for a specific holiday entry. If an entry on the Holiday Table screen matches with the current date and time, Communication Manager forwards the call to the first point that is defined in the coverage path. If there is no entry that matches with the current date and time, Communication Manager forwards the call to the subsequent point that is defined in the coverage path.
n	Communication Manager forwards the call to the subsequent point in the coverage path.

Holiday Table

Available only when **Holiday Coverage** is set to y for inside or outside calls.

The number of the holiday table used for holiday coverage.

Hunt After Coverage

Valid Entry	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned to the station of the last coverage point.
n	Coverage treatment is terminated. The call is left at the last available location, the principal or coverage point.

Linkage

One or two additional coverage paths in the coverage path chain.

Next Path Number

Valid Entry	Usage
1 to 9999	The number of the next coverage path in a coverage path chain. If the coverage criteria of the current coverage path is dissatisfied, the system checks in this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call stays out of coverage.
blank	The only path for the principal.

COVERAGE CRITERIA

Active

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call if at least one call appearance is busy.
n	The system does not redirect the call.

Busy

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call if all call appearances that accept incoming calls are busy.
n	The system does not redirect the call.

Don’t Answer

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call when the specified number of rings have been exceeded.
n	The system does not redirect the call.

All

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects all calls to coverage. This option overrides any other criteria. Calls redirect immediately to coverage. Overrides any other criteria administered for this field.
n	The system does not redirect the call.

DND/SAC/Go to Cover

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid entry	Usage
y	With this option, a calling user, when calling to another internal extension, can redirect a call immediately to coverage by pressing the **Go to Cover** button. A principal user can temporarily direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the **Send All Calls** or **Do Not Disturb** button. With the **Send All Calls** button, covering users can temporarily remove their telephones from the coverage path. **Note:** You must assign this criteria before a user can activate **Do Not Disturb (Hospitality Services)**, **Send All Calls (SAC)**, or **Go to Cover** features.
n	The system does not redirect the call.

Logged off/PSA/TTI

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

The system displays this field only when you set the **Criteria for Logged Off/PSA/TTI Stations** field to y.

Valid Entry	Usage
y	The system redirects the call after the number of rings exceeds the value specified in the **Number of Rings** field. The system displays the associated **Number of Rings** field only when the **Logged off/PSA/TTI** field is set to y.
n	The system does not redirect the call.

Number of Rings

Valid Entry	Usage
1 to 99	The number of times a telephone rings before the system redirects the call to the first point in the coverage path. By default, the value is 2.

COVERAGE POINTS

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially without steps beginning with Point 1. Each path can have up to six coverage points.

Subsequent coverage points should be unlisted if calls are redirected to:

- Message Center, a special Uniform Call Distribution hunt group
- Voice messaging
- The attendant

These calls normally queue and never redirect to another coverage point. Calls to hunt group queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full, or is nonexistent.

If the Coverage of Calls Redirected Off-Net feature is not supported, a remote coverage point functions as the last point in the coverage path because the system can no longer control calls once they redirect off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, calls redirected off-net can be monitored by the system and brought back for call coverage processing.

Valid Entry	Usage
extension	Redirects the call to an internal extension or announcement. **Note:** If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to h999	Redirects the call to the corresponding hunt-group, for example, `h32` routes to hunt group 32.
c1 to c750 c1 to c1000 (S8300D/duplex Media Servers)	Redirects the call to the corresponding coverage answer group, for example, `c20` routes to call coverage answer group 20.
r1 to r999 r1 to r1000 S8300D/duplex (Media Servers)	Redirects the call to the corresponding remote coverage point number, for example, `r27` routes to remote coverage point 27.
v + extension	Redirects the call to the corresponding Vector Directory Number (VDN) extension, for example, `v12345` routes to the VDN associated with extension 12345. **Note:** A VDN can be used only as the last administered point in a coverage plan.
y + extension	Redirects the call to an internal extension, announcement, or the corresponding Vector Directory Number (VDN) extension as per the current date and time set in Holiday Table.

Rng

Valid Entry	Usage
1 to 99 blank	The number of rings at this coverage point before the system redirects the call to the next point in the coverage path.

Terminate to Coverage Pts. with Bridged Appearances

Valid Entry	Usage
y	If activated, a call can alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Chapter 33: Crisis Alert System Parameters

Defines the system parameters associated with sending crisis alert messages.

Example command: `change system-parameters crisis-alert`

ALERT STATION

Every User Responds

Controls who needs to respond to a crisis alert.

Valid Entries	Usage
y	All users who have a crisis alert button are notified and must clear the alert for every emergency alert. Crisis alert buttons should be assigned only to attendant consoles and stations that must be notified of an emergency call.
n	All users are notified, but only one user needs to acknowledge an alert. This user might be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

ALERT PAGER

Alert Pager

Supports or restricts use of the Crisis Alert to a Digital Pager.

Crisis Alert Code

The first three digits in the crisis alert pager message. This should be the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

Required when Crisis Alert to a Digital Pager is enabled.

Related Links

Alert Pager on page 169

DTMF Duration - Tone (msec)

Available only when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
20 to 2550 (in increments of 10)	The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit.

Related Links

Alert Pager on page 169

Main Number

Identifies the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and excluded from the message sent to the pager. This entry is the last group of digits displayed in the pager message. Accepts up to 15 digits. Available only when Crisis Alert to a Digital Pager is enabled.

Related Links

Alert Pager on page 169

Originating Extension

Required when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
1 to 13	Requires a valid unassigned extension according to the dial plan

Related Links

Alert Pager on page 169

Pager Number

The number which Communication Manager dials to make the call to the pager. The ARS access code is automatically pre-pended to the **Pager Number** before Communication Manager dials the call.

Pause (msec)

Available when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
20 to 2550 (in increments of 10)	The length of time between DTMF tones for each digit.

Related Links

Alert Pager on page 169

Pin Number

If the pager service requires you to submit a PIN number, type the number in this field. The PIN number can be up to 15 digits. You can also enter the p (pause), # and * characters. A pause of up to 2 seconds can be used for the timing of the message. For instance, a pause might be necessary to allow time for the pager service to set up the correct pager message box.

The system displays this field only if Crisis Alert to a Digital Pager is enabled.

Related Links

Alert Pager on page 169

Retries

Available when Crisis Alert to A Digital Pager is enabled.

Valid Entry	Usage
0 to 10	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Related Links

Alert Pager on page 169

Retry Interval (sec)

Available when the Crisis Alert to a Digital Pager is enabled and **Retries** is set to a value from 1 to 10.

Valid Entry	Usage
30 to 60	The time in seconds between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

Related Links

Alert Pager on page 169
Retries on page 171

Chapter 34: CTI Link

Available if either ASAI Link Core Capabilities or Computer Telephony Adjunct Links are enabled on the system.

Example command: `add cti-link n`, where *n* is the CTI link number.

Related Links

[ASAI Link Core Capabilities](#) on page 795
[Computer Telephony Adjunct Links](#) on page 797

CTI link: page 1

```
add cti-link 1                                          Page   1 of   3
                                  CTI LINK
 CTI Link: 1
Extension:
     Type: ASAI
     Port:                                                    COR: 1
     Name:




BRI OPTIONS
               XID? y      Fixed TEI? n
       MIM Support? n
                          CRV Length: 2
```

COR

Class of Restriction (COR) number with the required restriction.

CTI Link

The Computer Telephony Integration (CTI) link number.

Extension

The extension for this link.

Name

The name associated with this CTI link.

Port

For an ASAI or ADJLK CTI link, the seven characters that specify a port.

Valid Entry	Usage
01 to 64	First and second numbers are the cabinet number
A to E	Third character is the carrier
01 to 20	Fourth and fifth characters are the slot number
01 to 32	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH.

Type

Valid Entry	Usage
ADJLK ADJ-IP ASAI ASAI-IP	The CTI link type.

BRI OPTIONS

CRV Length

Available for ASAI or ADJLK CTI links.

Valid Entry	Usage
1 to 2	The length of CRV for each interface

Fixed TEI

Indicates whether or not the endpoint has a fixed Terminal Endpoint Identifier (TEI). TEIs are administered for fixed TEI terminals. Available for ASAI or ADJLK CTI links.

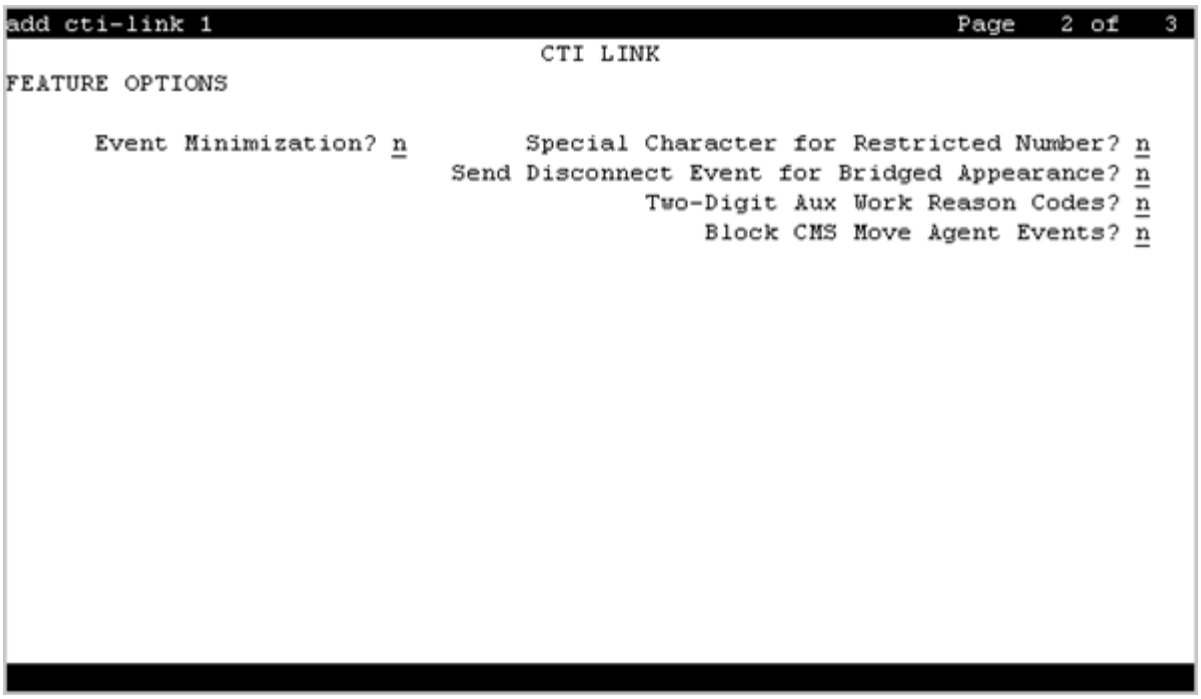
MIM Support

Indicates if Management Information Message (MIM) support is enabled for an ASAI or ADJLK link.

XID

For an ASAI or ADJLK CTI link, identifies Layer 2 XID testing capability.

CTI link: page 2



Comments on this document? infodev@avaya.com

Block CMS Move Agent Events

Enables or disables the blocking of certain event report messages involved with the move of agents while staffed. If the Call Management System sends an agent-move-while-staffed message (MVAGSFD8), ASAI does not send the associated agent Logout Event Report (C_Logout), Login Event Report (C_login) and Agent Work Mode Change event report messages.

Event Minimization

Enables or disables event minimization for this link that limits the number of event reports sent to an adjunct. This option can be used when event reports normally would be sent on multiple associations, but the adjunct can see only a single report. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link. When activated, only a single such event can be sent. The selection of the association on which the event is sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled).

Remove '+' from SIP Numbers

If you set the **Remove '+' from SIP Numbers** field on the CTI Link screen to y and if a leading plus (+) sign is present in a calling or called number sent over a CTI link, Communication Manager removes the leading plus (+) sign from the number. The default value is y.

Send Disconnect Event for Bridged Appearance

Indicates whether or not an event report is sent when a bridged appearance disconnects.

Special Character for Restricted Number

Enables or disables an ASAI CTI link that indicates the calling number restricted presentation within an event report. When enabled, a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then “*” is appended to the calling party number in the ASAI message.

Two-Digit Aux Work Reason Codes

Use this field to send a two-digit Reason code over the ASAI link. All messages that include the Aux-work Reason code allow codes from 1 to 99.

Related Links

Two-Digit Aux Work Reason Codes on page 328

Chapter 35: Data Module

Example command: `change data-module n`, where *n* is the module number.

Data module: page 1

BCC

Indicates the value that corresponds to the speed setting of the data module. Used with Data Line, Netcon, Processor Interface, Point-to-Point Protocol, Processor/Trunk (pdm selection), and System Port Data Modules. This field can be compared with the BCC value in an associated routing pattern when attempted calls utilizing the data module fail to complete. The BCC values must be the same.

Available only when ISDN-PRI or ISDN-BRI trunks are enabled.

Valid Entry	Usage
1	Relates to 56-bkps
2, 3, 4	Relates to 64 kbps

Related Links

ISDN-PRI on page 801
ISDN-BRI Trunks on page 801

Connected to

Used with Data Line and Processor/Trunk (pdm selection) Data Modules.

Valid Entry	Usage
dte	The Asynchronous Data Unit (ADU) is connected to Data Terminal Equipment.
isn	The ADU is connected to the Information Systems Network.

COS

The system does not display this field for ethernet.

Valid entry	Usage
0 to 15	The COS for the data module

COR

The system does not display this field for ethernet.

Valid entry	Usage
0 to 995	The COR for the data module.

Data Extension

Use this field to assign an extension to the data module. You can enter up to five digits. This value must conform to the dial plan.

ITC

The Information Transfer Capability (ITC) is used with 7500, Announcement, data-line, Netcon, Processor/ Trunk (pdm selection), Processor Interface, and System Port Data Modules. Indicates the type of transmission facilities used for ISDN calls originating from this endpoint. Unavailable for voice-only or BRI stations.

Valid Entry	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1’s density digital transmission. In other words, a sequence of eight digital zeros is converted to a sequence of seven zeros and a digital one.
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1’s density digital transmission. In other words, digital information is sent exactly as is.

Name

The name of the user associated with the data module. The name is optional and can be blank. Accepts up to 27 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Port

A port location to which the data module is connected. Used with 7500, Data Line, Ethernet, Processor/Trunk, PPP, System Port, and World Class BRI Data Modules.

Characters	Meaning	Value
1 to 2	Cabinet number	1 to 64
3	Carrier	A to E
4 to 5	Slot number	0 to 20
6 to 7	Circuit Number	• 01 to 31 • 01 to 16 • 01 to 08 17/33

Note:

An x in the **Port** field indicates that there is no hardware associated with the port assignment, also known as Administration Without Hardware (AWOH). These stations are called phantom stations. If this data module is designated as a secondary data module, you cannot enter x. The port of a primary data module cannot be changed to x if a secondary data module is administered.

Remote Loop-Around Test

Used with Processor/Trunk Data Modules. Available with a pdm or tdm type trunk.

Valid Entry	Usage
y	The data module supports a loop-back test at the EIA interface. Avaya equipment supports this loop-back test which is not required by Level 2 Digital Communications Protocol.
n	Abort a request for this test

Related Links

Type on page 181

Secondary data module

Used with Processor/Trunk Data Modules. Available with a pdm type trunk. The primary data module must be administered before the secondary data module can be added. A data module administered without port hardware cannot be the secondary data module.

Valid Entry	Usage
y	This PDM is the secondary data module used for Dual I-channel AUDIX networking.
n	This is the primary PDM, or this data module is not used for AUDIX networking.

Related Links

Type on page 181
Port on page 180

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Type

The type of data module.

Valid Entry	Usage
7500	Assigns a 7500 Data Module. The 7500 data module supports automatic TEI, B-channel, maintenance and management messaging, and SPID initialization capabilities. Both voice and data BRI endpoints are assigned to either the ISDN-BRI - 4-wire S/T-NT Interface circuit pack or the ISDN-BRI - 2-wire U circuit pack. Each can support up to 12 ports. Since BRI provides multipoint capability, more than one ISDN endpoint (voice or data) can be administered on one port. For BRI, telephones having SPID initialization capabilities support multipoint administration. Multipoint administration can only be activated if endpoints administered on the same port is not a fixed tie endpoint and station on the same port does not have B-channel data capability. Currently, multipoint is restricted to two endpoints per port.
announcement	Assigns an announcement data module. The announcement data module is built-in to the integrated announcement circuit pack. The system uses this data module to save and restore the recorded announcements file between the announcement circuit pack and the system memory.
data-line	Assigns a Data Line Data Module. Administered ports on the Data Line circuit pack (DLC) can connect EIA 232C devices to the system. The DLC, with a companion Asynchronous Data Unit (ADU), provides a less expensive data interface to the system than other asynchronous DCP data modules.

Table continues…

Valid Entry	Usage
	The DLC supports asynchronous transmissions at speeds of Low and 300, 1200, 2400, 4800, 9600, and 19200 bps over two-pair full-duplex lines. These lines can have different lengths, depending on the transmission speed and wire gauge. The DLC has eight ports. The connection from the port to the EIA device is direct, meaning that no multiplexing is involved. A single port of the DLC is equivalent in functionality to a data module and a digital line port. The DLC is displayed as a data module to the Digital Terminal Equipment (DTE) and as a digital line port to the server running Communication Manager. The DLC connects the following EIA 232C equipment to the system: • printers • non-intelligent data terminals • intelligent terminals, personal computers (PCs) • host computers • Information Systems Network (ISN), RS-232C local area networks (LANs), or other data switches
ethernet	Assigns an Ethernet data module. A 10BaseT port can be administered on the Control-LAN (C Lan) circuit pack. This port provides a TCP/IP connection to network hub or LAN.
ni-bri	Assigns an NI-BRI Data Module.
pdm	Assigns a DCE interface for Processor/Trunk Data Modules. A Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs) can be administered. The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface for connection to equipment such as data terminals, CDR output devices, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. It also provides a Digital Communications Protocol (DCP) interface to the digital switch. DCE is the equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel. The MTDM provides an Electronic Industries Association (EIA) Data Terminal Equipment (DTE) interface for connection to off-premises private line trunk facilities or a switched telecommunications network and a DCP interface for connection to the digital switch. DTE is the equipment comprising the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host computer, the terminal, the host, and their associated modems or data modules make up the DTE. The MTDM or 7400A Data Module also can serve as part of a conversion resource for Combined Modem Pooling.
ppp	Assigns a Point-to-Point Protocol data module. A synchronous TCP/IP port can be administered on the C-Lan circuit pack. These ports are tailored to provide TCP/IP connections for use over telephone lines.
system-port	Assigns a System Port Data Module.

Table continues…

Valid Entry	Usage
tdm	Assigns a DTE interface for Processor/Trunk Data Modules. See the pdm entry above.
wcbri	Assigns a World Class BRI Data Module.

ABBREVIATED DIALING

List1

The abbreviated dialing list for the data module. Used with 7500, Data Line, Netcon, Processor/ Trunk, Processor Interface, and World Class BRI Data Modules. Supports Data Hot Line. This field can be left blank.

Valid Entry	Usage
e	Enhanced
g	Requires a group list number.
p	Requires a personal list number.
s	System

ASSIGNED MEMBER

Ext and Name

The extension number and name of the previously administered user with associated Data Extension buttons, who shares the module. Used with Data Line, Announcement, Netcon, Processor/Trunk, Processor Interface, and System Port Data Modules.

SPECIAL DIALING OPTION

Identifies the type of dialing for calls when this data module originates calls. Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules.

Valid Entry	Usage
hot-line	When the user goes off-hook on the data module, the hot line destination number gets dialed.
default	When the user goes off-hook on the data module and presses **Enter** at the DIAL prompt, the default dialing destination number gets dialed.
blank	Normal keyboard dialing

Data module: page 2: Type data-line

CAPABILITIES

Busy Out

Enables or disables the placement of the DLC port in a busied-out state once the DTE control lead to the DLC is dropped. This option should be enabled for DTEs that are members of a hunt group and to activate busy out when DTE is turned off, so that calls do not terminate on that DTE.

Configuration

Activates or deactivates the viewing and changing of options from the DTE. Available only when **KYBD Dialing** is enabled. This option normally is enabled for originate/ receive DTE such as non-intelligent terminals and disabled for intelligent devices such as computers.

Related Links

KYBD Dialing on page 184

KYBD Dialing

Note:

ADU-type hunt groups connecting the system to terminal servers on a host computer should have these hunt group extensions assigned as no keyboard dialing.

Valid Entry	Usage
y	Enables keyboard dialing, allowing data endpoints to originate calls using the EIA 232C interface and obtain ASCII feedback text. The user gets the dial prompt. This option normally is enabled for a originate/receive DTE that has a need to set up data calls.
n	Disables keyboard dialing. Calls do not originate at the DTE and the system does not display the text feedback during call setup or cancellation. Data call answering is still supported but without text feedback. If the **Low** speed setting is enabled, keyboard dialing should be disabled.

OPTIONS

Answer Text

Available only with **KYBD Dialing**.

Applies to the following messages:

- INCOMING CALL
- ANSWERED

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- DISCONNECTED
- DISCONNECTED OTHER END

Valid Entry	Usage
y	Text messages can be delivered to the DTE when a call is being answered. Applies to DLC-generated text as well as text received from the system.
n	The system still generates the text, but the DLC prevents it from being sent to the device. Usually is disabled when the answering DTE is a computer or an intelligent device.

Related Links

KYBD Dialing on page 184

Connected Indication

Available only with **KYBD Dialing**.

Valid Entry	Usage
y	Generates a “CONNECTED” message to the DTE when the connection has been established.
n	The connected indication is provided by the DLC activating its EIA 232C control lead.

Related Links

KYBD Dialing on page 184

Dial Echoing

Available only with **KYBD Dialing**.

Valid Entry	Usage
y	Echos characters back to the DTE.
n	Disables Dial Echoing. Dial Echoing should be disabled when keyboard dialing is done by an intelligent device.

Related Links

KYBD Dialing on page 184

Disconnect Sequence

Selects the sequence for a disconnect. Available only with **KYBD Dialing**.

Valid Entry	Usage
long-break	A long-break is greater than 2 seconds.
two-breaks	Two-breaks is within 1 second.

Related Links

KYBD Dialing on page 184

Parity

Valid Entry	Usage
even odd mark space	The type of parity. The DLC generates the parities when call setup text is sent to the DTE. The DLC ignores the parity when receiving dialing characters. Parity has no bearing on the far end. It is used by the DLC-to-terminal communications during call setup. Available only with **KYBD Dialing**.

Related Links

KYBD Dialing on page 184

Permit Mismatch

Enables or disables the Permit Mismatch feature. Permit Mismatch:

- The EIA interface can operate at a rate different than that agreed to in the data module handshake. The data module handshake is always the highest compatible rate as determined by the reported speed option of each data module.
- Instructs the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module. The DLC reports the highest-optioned speed, all the lower speeds, or the previously-selected auto-adjust speed during the handshake process.
- Eliminates the need to change the DTE/DLC speed every time a call is placed to and from an endpoint operating at a different speed.

Caution:

You must be cautious while sending information from a DTE/ DCE that is transmitting data at higher rates than that of the far end. Sustained usage of this type of transmission might result in data loss. When you enable this option, the DTE must match the highest speed selected for the associated DLC port.

This option is meant for use by a DTE device operating locally at a higher baud rate than that of its far-end connection, but transmitting relatively low amounts of data, for example, a user typing at a terminal.

Note:

The Low speed setting is not reported as an available speed when **Permit Mismatch** is enabled.

Related Links

SPEEDS on page 186

SPEEDS

Enables or disables the following operating speeds:

Valid Entry	Usage
Low	Instructs the DLC to operate at a low speed from 0 to 1800 bits per second (bps). Disable if **KYBD Dialing** is enabled.
300, 1200, 2400, 4800, 9600, or 19200	The DLC can be any one of these speeds. The speed is matched for the duration of the call, from call setup to call takedown. For multiple speeds, select three or more entries. When you select multiple speeds and disable autoadjust, the DTE speed must be the highest selected speed. This is required because all feedback text is delivered to the DTE at the highest selected speed.
Autoadjust	• Tells the DLC port to automatically adjust to the operating speed and to the parity of the DTE it is connected to • Can be used with any of the speeds listed • Applies only to calls originated by the user through Keyboard Dialing • Available only when **KYBD Dialing** is enabled

Related Links

KYBD Dialing on page 184

Data module: page 2 - Type 7500, WC-BRI, NI-BRI

BRI LINK/MAINTENANCE PARAMETERS

Endpt Init

Indicates whether the terminal endpoint has initialization capability. Endpoint initialization is a procedure required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made through a Service Profile Identifier (SPID) administered into the system and entered into the ISDN-BRI terminal. For a ISDN-BRI terminal to become operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. This means that the SPID of the new or reused terminals must be programmed to match the administered SPID value. Used with 7500, World Class BRI, and NI-BRI Data Modules.

Fixed TEI

Indicates whether an endpoint has Fixed Terminal Equipment Identifier (TEI) capability. TEI identifies a unique access point within a service. For Fixed TEI stations, the TEI must be administered. For terminals with automatic TEI capability, the associated TEI is assigned by the system. Used with 7500, World Class BRI, and NI-BRI Data Modules.

MIM Mtce/Mgt

Enables or disables Management Information Message (MIM) support. MIM provides terminal support for MIM Maintenance and Management capabilities, other than endpoint initialization. Used with 7500 Data Modules.

MIM Support

Enables or disables the capability of MIM endpoint initialization (SPID support), and other Maintenance/Management. Used with 7500 Data Modules.

SPID

The Service Profile Identifier (SPID) is a variable parameter of up to 10 digits. The SPID must be different for all terminals on the ISDN-BRI and from the Service SPID. The SPID should always be assigned. If the SPID is unassigned for the first ISDN-BRI on a port, any other ISDN-BRI assignment to that port is blocked. Used with 7500, World Class BRI, and NI-BRI Data Modules. Available only if **Endpt Init** is enabled.

Related Links

Endpt Init on page 187

TEI

Available only if **Fixed TEI** is enabled.

Valid entry	Usage
0 to 63	The Terminal Endpoint Identifier (TEI) is a layer 2 addressing parameter used by Communication Manager to exchange information with BRI endpoints over the point-to-point signaling link. Used with 7500, World Class BRI, and NI-BRI Data Modules.

Related Links

Fixed TEI on page 187

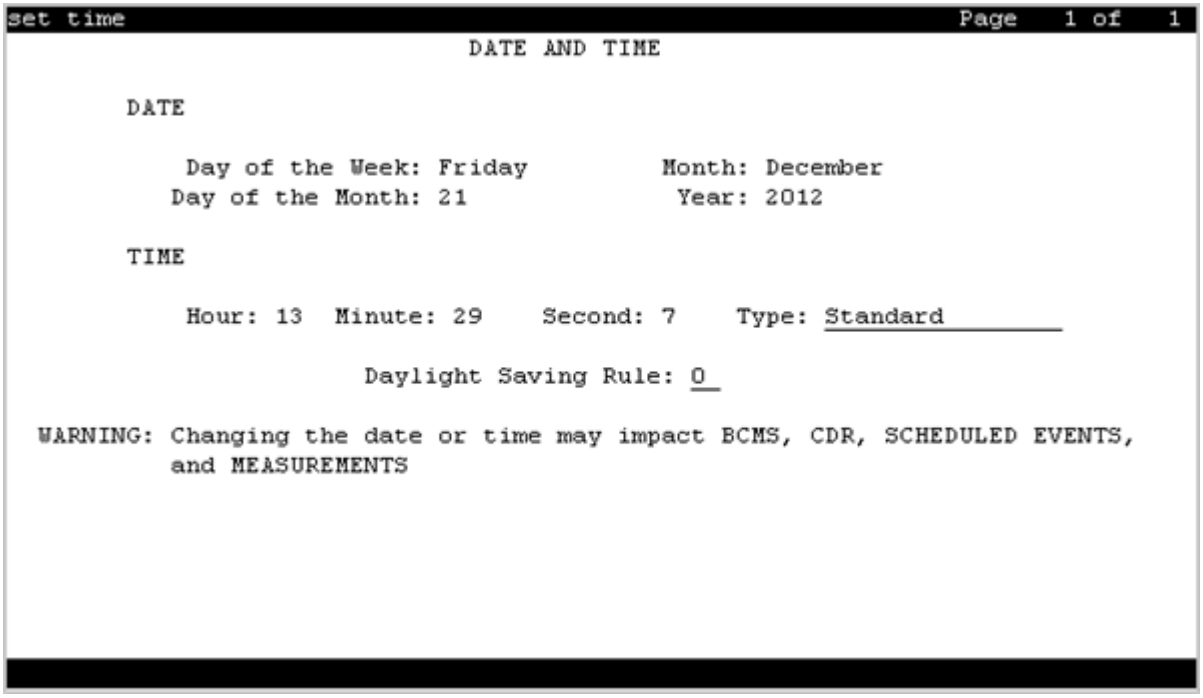
XID

Enables or disables layer 2 Exchange identification (XID) testing capability. Used with 7500, World Class BRI, and NI-BRI Data Modules. In almost all cases, disable the field.

Chapter 36: Date and Time

Displays the system date and time, selects the daylight saving plan number, if any, and shows whether the current time is standard time or daylight saving. Settings on this screen affect the internal clock and timestamp of the server running Communication Manager. Update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until you enter the date and time.

Example command: `set time`



Day of the Month

Valid entry	Usage
1 to 31	The current day of the month. The system clock uses this as the current date.

Day of the Week

Valid Entry	Usage
Sunday through Saturday	The current day of the week. The system clock uses this as the current day.

Daylight Saving Rule

Valid Entry	Usage
0 to 15	The daylight saving rule in use for the system. The system clock uses this as the current daylight saving rule.

Related Links

Daylight Saving Rules on page 192

Hour

Valid Entry	Usage
0 to 23	The current hour to be used by the system clock. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Minute

Valid Entry	Usage
0 to 59	The current minute. The system clock uses this as the current minute.

Month

Valid Entry	Usage
January to December	The current month. The system clock uses this as the current month.

Second

Displays the seconds and cannot be modified. Resets to zero when saved.

Type

Valid Entry	Usage
daylight-savings	Indicates daylight saving time is in effect.
standard	Indicates standard time is in effect.

Year

The current year in 20*XX* format. The system clock uses this as the current year.

Chapter 37: Daylight Saving Rules

Administers up to 15 customized daylight saving rules. Specifies the exact date and time each daylight saving rule goes into effect and when it stops. Rule 0 makes no adjustment to the system clock for daylight saving and cannot be modified.

In this example, Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after March 8 at 2:00 a.m. with a 01:00 increment. Daylight Saving Time stops on the first Sunday on or after November 1 at 2:00 a.m., also with a 01:00 increment used as a decrement when switching back to Standard time. Telephone displays reflect these settings.

```
1:    Start: first _ Sunday_ on or after _March   8_ at 2:00_  01:00__
    Stop:  first _ Sunday_ on or after November 1_ at 2:00_
```

Example command: `change daylight-savings-rules`

```
change daylight-savings-rules                                Page   1 of   2
                        DAYLIGHT SAVING RULES

Rule          Change Day                  Month  Date   Time     Increment

 0: No Daylight Saving

 1: Start: first Sunday___  on or after March___  8_   at 02:00   01:00
    Stop: first Sunday___  on or after November_  1_   at 02:00
 2: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
 3: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
 4: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
 5: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
 6: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
 7: Start: first _________  on or after _________ __   at __:__   __:__
    Stop: first _________  on or after _________ __   at __:__
```

Change Day (Start)

Valid Entry	Usage
Sunday to Saturday	The day of the week the clock moves ahead to begin daylight saving.
Day	The clock changes on the exact date entered for **Month (Start)** and **Date (Start)** values.

Related Links

Month (Start) on page 194

Change Day (Stop)

Valid Entry	Usage
Sunday to Saturday	The day of the week the clock moves back to standard time
Day	The clock changes on the exact date entered for the **Month (Stop)** and **Date (Stop)** values

Related Links

Month (Stop) on page 194

Date (Start)

Valid Entry	Usage
0 to 31	The day of the month the clock moves ahead to begin daylight saving.

Date (Stop)

Valid Entry	Usage
0 to 31	The day of the month the clock moves back to standard time

Increment (Start)

Valid Entry	Usage
0 to 23	The number of hours the clock moves ahead for daylight saving and moves back to return to standard time.
0 to 59	The number of minutes you want the clock to move ahead for daylight saving and to move back to return to standard time.

Month (Start)

Valid Entry	Usage
January to December	The month the clock moves ahead to begin daylight savings

Month (Stop)

Valid Entry	Usage
January to December	The month the clock moves back to standard time

Rule

The daylight saving rule number.

Time (Start)

Valid Entry	Usage
0 to 23	The hour the clock moves ahead to begin daylight saving. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.
0 to 59	The minute the clock moves ahead to begin daylight saving.

Time (Stop)

Valid Entry	Usage
0 to 23	The hour the clock moves back to standard time. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.
0 to 59	The minute the clock moves back to standard time

Chapter 38: DCS to QSIG TSC Gateway

Determines when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. Maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the QSIG MWI-Prefix screen. Assigns the voice mail number used when a DCS served-user node interrogates a QSIG message center.

Available only if **Interworking with DCS** is enabled for the system.

Example command: `change isdn dcs-qsig-tsc-gateway`

Related Links

Interworking with DCS on page 814

AAR/ARS Access Code

The AAR/ARS Access Code. Accepts up to four digits, including characters * and #.

Mach ID

Valid Entry	Usage
digits	A unique machine ID. Do not repeat a machine ID if it is already associated with a processor channel on an Ethernet link. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Sig Grp

Valid Entry	Usage
digits	The assigned signaling group number for each machine ID. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

TSC Index

The TSC Index for each machine ID.

Valid Entry	Usage
1 to 64	The assigned signaling group number for the qsig-mwi application type.

Voice Mail Number

The complete Voice Mail Dial Up number. Accepts up to 17 digits.

Chapter 39: Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. It determines the beginning digits and total length for each type of call that Communication Manager needs to interpret. The Dial Plan Analysis Table and the Dial Plan Parameters screen work together to define the system's dial plan.

Example command: `change dialplan analysis`

Call Type

Valid Entry	Usage
aar	Automatic Alternate Routing — Routes calls within a company's own private network. Requires that **ARS/ AAR Dialing without FAC** is enabled.
ars	Automatic Route Selection — Routes calls that go outside a company over public networks. ARS also routes calls to remote company locations without a private network. Requires that **ARS/ AAR Dialing without FAC** is enabled.
attd	Attendant — Defines how users call an attendant. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code. The attendant access code can also be administered by entering an fac or dac.
dac	Dial access code — Using this feature, you can use trunk access codes (TAC) and feature access codes (FAC) in the same range.
enb-ext	Enbloc extension — Defines a block of extensions that must be dialed using a prefix when the caller dials from a keypad. These extensions can be dialed without a prefix if the caller dials enbloc, for example, from a station call log.
ext	Primary extension — Defines extension ranges that can be used on the system. Extension can have a first digit of 0 through 9 and can be one to seven digits in length. Extension cannot have the same first digit as a 1-digit ARS or AAR feature access code (FAC).
fac	Feature access code only — Users dial an FAC instead of programming a button.
pext	Prefixed extension — Identifies the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits make up the extension number and are then processed. With a prefixed extension, extensions numbers with any dialed string can be used. The extension length must be specified on the table.
udp	Uniform Dial Plan — Shares a common dial plan among a group of servers.

Related Links

ARS/AAR Dialing without FAC on page 794

Dial Access on page 481
UDP Extension Search Order on page 204

Dialed String

The digits that Communication Manager analyzes to determine how to process the call. Two Dial Plan entries can use the same Dialed String only if the Dialed String consists of one digit. Longer Dialed Strings must all be unique. A new entry cannot be administered if it causes an existing extension, feature access code, or trunk access code to become inaccessible.

Call Type	Valid Entry	Usage
aar	0 to 9	Dialed String entry can be one to six digits long.
ars	0 to 9	Dialed String entry can be one to six digits long.
attd	0 to 9	Dialed String entry can be one or two digits long.
dac	0 to 9 * #	Dialed String can be one to four characters long. The characters “*” and “#” can only be used as a first digit.
enb-ext	0 to 9	Dialed String can be one to ten digits long. The corresponding Total Length can be up to thirteen digits.
ext	0 to 9	Dialed String can be one to ten digits long. The extension cannot have the same first digit as a one-digit ARS or AAR feature access code (FAC). The corresponding Total Length can be up to thirteen digits.
fac	0 to 9 * #	Dialed String can be one to four characters long. The characters “*” and “#” can only be used as a first digit. A FAC must have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems might occur when, for example, three-digit FACs and four-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it must have the longest total length in the dial plan.
pext	0 to 9	The prefix is the first digit and an extension number of up to five digits in length. The maximum length of a prefix and extension combination is six digits long. A prefixed extension cannot have the same first digit as a dial access code.

Table continues…

Call Type	Valid Entry	Usage
		A prefixed extension cannot have the same dialed string as the ARS or AAR FAC.
udp	0 to 9	Dialed String can be one to ten digits long. The corresponding Total Length can be up to eighteen digits.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays **all**.

Related Links

Multiple Locations on page 804

Percent Full

Valid Entry	Usage
0 to 100	The percentage of system memory resources that have been allocated for the dial plan currently used. For details on the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Total Length

The number of digits for this call type. Must be greater than or equal to the number of digits in the Dialed String.

Valid Entry	Usage
1 to 2	attd
1 to 4	dac

Table continues…

Valid Entry	Usage
1 to 4	fac
1 to 13	ext
2 to 6	pext

Related Links

Dialed String on page 199

Chapter 40: Dial Plan Parameters

The Dial Plan Parameters screen works with the Dial Plan Analysis Table to define the system dial plan.

It also controls the appearance of digit extensions on station displays. These multi-digit extensions can be hard to read as a block. Using Communication Manager you can select the display format for 6- to 13-digit extensions.

AAR/ARS Internal Call Prefix

The digits entered here get concatenated with the calling or called extension. Accepts up to eight digits that excludes * or #. Available only when **ARS/AAR Dialing Without FAC** is enabled for the system. Requires administration of **AAR/ARS Internal Call Total Length**.

Related Links

AAR/ARS Internal Call Total Length on page 202

AAR/ARS Internal Call Total Length

Available only if **ARS/AAR Dialing Without FAC** is enabled for the system.

Valid Entry	Usage
6 to 10 blank	The total length of the internal call digit string, including the Internal Call Prefix and the calling or called extension. Requires administration of an **AAR/ARS Internal Call Prefix**.

Note:

The longest extension length on the Dial Plan Analysis table, plus the length of the **ARS/AAR Internal Call Prefix**, must be equal to or greater than the **ARS/AAR Internal Call Total Length** value.

ETA Node Number

Valid Entry	Usage
1 to 999 blank	The number of the destination server for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions outside the UDP table.

ETA Routing Pattern

Valid Entry	Usage
1 to 999	The number of the routing pattern to reach the destination server.

Local Node Number

Valid Entry	Usage
1 to 63	Identifies a specific node in a server network. Must match the DCS switch node number and the CDR node number if they are specified.
blank	Leave blank if automatic restoration, DCS, and CDR are unused.

Retry ARS/AAR Analysis if All-Location Entry Inaccessible

Available with Communication Manager Release 4.0.*x* or later.

Valid Entry	Usage
y	The system finds and uses the best possible entry in the per-location ARS table, if the all-location table points to a trunk group that cannot be accessed because the network has fragmented.
n	The system stops retry of ARS analysis when a trunk group cannot be accessed as a result of the network being fragmented.

UDP-ARS Calls Considered Offnet

Valid entry	Usage
y	If you set the **Net** field to ARS and the **UDP-ARS Calls Considered Offnet** field to y, the call is offnet. If UDP-ARS calls are off-net then: • Users with the COS setting Restrict Call Fwd-Off Net enabled cannot forward calls to a UDP-ARS number. • Stations that are outward-restricted (through the COR) cannot place calls to a UDP-ARS number. • Toll analysis is applied when a call is placed by a toll-restricted user to a UDP-ARS number. • COR-COR checks are inapplicable when a user dials a call to a UDP-ARS number.
n	If you set the **Net** field to ARS and the **UDP-ARS Calls Considered Offnet** field to n, the call is onnet.

For more information about the **Net** field, see Uniform Dial Plan Table section of the *Avaya Aura® Communication Manager Screen Reference*, 03-602878.

UDP Extension Search Order

Specifies the first table to search to match a dialed extension. If the dialed extension is not found in the specified place, the system searches for the dialed extension in the alternate place. Available if **Uniform Dialing Plan** is enabled for the system.

Valid Entry	Usage
local-extensions-first	Searches the local server first to match a dialed extension, and if not found, then uses the UDP tables to route the call.
udp-table-first	Searches the UDP tables for an off-switch conversion, and if not found, then searches the local server for the dialed extension.

EXTENSION DISPLAY FORMATS

Extension display format

Specifies how the system punctuates extensions for display. The punctuation field is divided into two columns, one for Inter-Location/SAT displays, and one for Intra-Location displays.

Comments on this document? infodev@avaya.com

In Communication Manager 6.0, the Inter-Location/SAT column is divided into two columns. You can insert a dial prefix in the Inter-Location column, if appropriate, before an enbloc extension. With this, called stations can store a dialable number in their call logs.

Note:

The maximum length of a displayed extension (including punctuation) is 13 characters. You must trade off a punctuation mark so that you can insert a prefix digit. You cannot insert a prefix digit or a punctuation mark into the inter-location format for a 13 digits extension.

Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (.) are used on SAT screens in place of blanks. The following table gives the maximum number of punctuation marks permitted for each extension length.

The number of punctuation marks that the system supports is determined by the number of “x”s in the format.

- If the format contains fewer than six x's, no punctuation marks can be entered.
- If the format contains six or more x's, the maximum number of punctuation marks is determined by the following table.

Extension Length	Maximum Punctuation Marks	Maximum Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid Entry	Usage
xx.xx.xx	Can contain all “x” characters (no punctuation) or a combination of “x” characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions display. A format must be specified or the default takes effect. This field cannot be left blank. The default values for the 8-, 9-, 10-, 11-, 12-, and 13-digit fields are shown in the following example.

```
EXTENSION DISPLAY FORMATS

                          SAT             Inter-Location  Intra-Location
       6-Digit Extension:   xx.xx.xx        xx.xx.xx        xx.xx.xx
       7-Digit Extension:   xxx-xxxx        xxx-xxxx        xxx-xxxx
       8-Digit Extension:   xx.xx.xx.xx     xx.xx.xx.xx     xx.xx.xx.xx
       9-Digit Extension:   xxx-xxx-xxx     xxx-xxx-xxx     xxx-xxx-xxx
      10-Digit Extension:   xxx-xxx-xxxx    xxx-xxx-xxxx    xxx-xxx-xxxx
      11-Digit Extension:   xxxx-xxx-xxxx   xxxx-xxx-xxxx   xxxx-xxx-xxxx
      12-Digit Extension:   xxxxxx-xxxxxx   xxxxxx-xxxxxx   xxxxxx-xxxxxx
      13-Digit Extension:   xxxxxxxxxxxxx   xxxxxxxxxxxxx   xxxxxxxxxxxxx
```


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Chapter 41: Digit Absorption

Implements up to five-digit absorption lists. Digit Absorption is required for each local telephone company central office (CO) and for a Foreign eXchange (FX) trunk group connected to a CO. Each outgoing digit string from the server running Communication Manager to the CO is treated according to entries in the Absorption Treatment Assignment section of the screen.

Available only if Digit Absorption is administered for the trunk group.

Example command: `change digit-absorption` *n*, where *n* is the absorption digit.

Related Links

Digits on page 882
Digit Absorption List on page 865

Absorption Treatment Assignment

Valid Entry	Usage
A to F	The chosen treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Absorption Treatment Information

Shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II.

List Number

Valid Entry	Usage
0 to 4	The Digit Absorption List number that is referenced from the associated trunk group.

Chapter 42: Display Parameters

Establishes how extensions of 6 to 13 digits are punctuated.

Example command: `change display-parameters`

Related Links

Extension display format on page 204

EXTENSION DISPLAY FORMATS

The fields in this section of the screen override similar fields on the Dial Plan Parameters screen. If you leave these fields blank, the values on the Dial Plan Parameters screen apply.

Default Call Appearance Display Format

Affects call appearances only on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances remain unaffected.

Valid Entry	Usage
inter-location	The complete extension on downloadable call appearance buttons. This is the default.
intra-location	A shortened version of the extension on downloadable call appearance buttons.

Display between Locations with these Display Parameters

The system displays this field only when you enable the **Multiple Locations** field. This field cannot be left blank.

Valid Entry	Usage
inter-location	Communication Manager uses the inter-location display format for calls between designated locations. This is the default value.
intra-location	Communication Manager uses the intra-location display format for calls within a location. You can use the intra-location format within a branch to display short extensions within that location. This Display Parameter lets you display short

Table continues…

Valid Entry	Usage
	extensions between two or more closely associated locations. For example, if you assign display parameter 3 to locations 7 and 17, and set the field to intra-location, Communication Manager uses intra-location format for calls between locations 7 and 17, however uses inter-location calls from 7 to any other location other than 17, and for calls from 17 to any other location other than 7.

Extension Display Format

Use this field to specify how the system must punctuate extensions for inter-location and intra-location displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (.) are used on SAT screens in place of blanks.

The number of punctuation marks that you can use is determined by the number of *x*s in the format:

- If the format contains fewer than six *x*s, there can be no punctuation marks.
- If the format contains six or more *x*s, the maximum number of punctuation marks is determined by the following table.

Extension length	Maximum punctuation marks	Maximum total length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13

Valid entry	Usage
xx.xx.xx blank	For six or more *x*s, characters can contain: • All *x* characters without punctuation • A combination of *x* characters and up to two hyphens • Spaces • Periods

Inter-Location

Specifies punctuation for calls between locations. This is the default.

Intra-Location

Specifies punctuation for calls within a location.

Chapter 43: DS1 Circuit Pack

Administers all DS1 circuit packs.

Example command: `add ds1 n`, where *n* is the board location.

DS1 Circuit Pack: page 1

Bit Rate

Note:

TN464C and later release circuit packs have an option switch that must be set to match this **Bit Rate** value.

Valid Entry	Usage
1.544	The maximum transmission rate for DS1 circuit packs that support T-1 service.
2.048	The maximum transmission rate for DS1 circuit packs that support E-1 service.

Channel Numbering

The ETSI and ISO QSIG specifications require that B-channels on an E1 be encoded as 1 to 30 in the Channel ID IE. Prior to the existence of this field, Avaya Communication Manager only used this scheme for Country Protocols 2a (Australia) and 13a (Germany 1TR6). Available only with ISDN-PRI signaling on a private network. The interface must be peer master or peer slave.

2.048 bit rate options:

- timeslot
- sequential

  If Communication Manager is connected via QSIG trunks to a switch or server supporting the ETSI QSIG or ISO QSIG specifications, this field must be sequential.

Related Links

Signaling Mode on page 223

Bit Rate on page 210
Connect on page 211
Interface on page 214

Connect

To control communications at layers 2 and 3 of the ISDN-PRI protocol, this field to specifies what is on the far end of this DS1 link.

Available only for ISDN-PRI signaling.

Valid Entry	Usage
pbx	The DS1 link is connected to another switch in a private network.
line-side	Communication Manager is acting as the network side of an ISDN-PRI interface. Used to connect to Roll About Video equipment.
network	The DS1 link connects Communication Manager to a local telephone company central office or any other public network switch.
host	The DS1 link connects Communication Manager to a computer.

Related Links

Signaling Mode on page 223

Country Protocol

The country protocol used by the far-end server. For connections to a public network, your network service provider can tell you which country protocol they are using.

Available only with ISDN-PRI and CAS signaling.

Valid Entry	Usage
1 to 25	The country protocol used by the local telephone company central office at which this link terminates.
etsi	The network service provider uses the European Telecommunications Standards Institute (ETSI) protocol and the **Signaling Mode** is isdn-pri.

Related Links

Signaling Mode on page 223
Country options table on page 783

CRC

Indicates whether a cyclic redundancy check (CRC) will be performed on transmissions that the DS1 circuit pack receives.

Valid Entry	Usage
y	The **Signaling Mode** is CAS and the DS1 link is providing E-1 service.
n	All other applications.

Related Links

Signaling Mode on page 223

D-Channel

Available only with a Japanese 2-Mbit trunk circuit pack and the ISDN-PRI **Signaling Mode**.

Valid Entry	Usage
1 to 31	The Japanese 2-Mbit trunk circuit pack, when administered to support ISDN-PRI signaling, assigns the D-channel to any channel from 1 to 31 in an E-1 facility.

Related Links

Signaling Mode on page 223

DCP/ANALOG Bearer Capability

Sets the information transfer capability in a bearer capability IE of a setup message to speech or *3.1kHz*. Available only with the ISDN-PRI **Signaling Mode**.

Valid Entry	Usage
3.1kHz	Provides 3.1 kHz audio encoding in the information transfer capability.
speech	Provides speech encoding in the information transfer capability.

Related Links

Signaling Mode on page 223

Disable Restarts

Controls whether outgoing RESTART messages are sent. Also used to disable QSIG restarts. Available when:

- Country Protocol is 3 (Japan)

- Country Protocol is ETSI
- Peer Protocol is QSIG

Valid Entry	Usage
y	Outgoing restarts are disabled. In other words, RESTART messages are not sent.
n	Outgoing RESTART messages are sent. This is the default.

DMI-BOS

The DMI/BOS protocol is used for high-speed digital communications between a host computer and Communication Manager. With this 24-channel protocol, channels 1 to 23 of the DS1 link carry data and channel 24 carries control signaling. DMI/BOS has greater capacity than a robbed-bit 24-channel facility. Available only when **Signaling Mode** is common-chan.

Valid Entry	Usage
y	Activates the Digital Multiplexed Interface-Bit Oriented Signaling (DMI-BOS) format.
n	Uses an Avaya proprietary format.

Related Links

Signaling Mode on page 223

Framing Mode

Selects either superframe or extended superframe for T1 service on the DS1 link. The framing mode must match the mode used on the other end of the link. Available only with T1 service.

Tip:

Use ESF when the service provider supports it, especially if the facility gets upgraded to ISDN. The ESF format provides enhanced performance measurements and uses a sophisticated error-checking method to ensure data integrity.

Valid Entry	Usage
d4	The basic DS1 superframe. Use this mode only for voice traffic.
esf	The extended superframe format. Use this mode for digital data traffic. A TN464F, TN767E, or a later suffix DS1 circuit pack requires the administration of ESF Data Link options.

Related Links

Bit Rate on page 210
ESF DATA LINK OPTIONS on page 227

Idle Code

Caution:

The **Country Protocol** sets the default idle code. Do not change the default without assistance from Avaya or your network services provider.

Valid Entry	Usage
Any 8-digit string of 0's and 1's	Sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch or server.

Related Links

Country Protocol on page 211

Interconnect

For E1 service using channel-associated signaling, tells Communication Manager whether the DS1 circuit pack is using a public or private network protocol. This value must agree with the **Group Type** value administered for the trunk group.

Available only if the **Signaling Mode** is CAS.

Valid Entry	Usage
pbx	The board operates as a tie trunk circuit pack.
CO	The board operates as a local telephone company central office (CO) or DID circuit pack. Use for Enterprise Mobility User (EMU)/EC500 administration.

Related Links

Signaling Mode on page 223
Group Type on page 482

Interface

Controls how the server negotiates glare with the far-end switch. The servers at either end of the DS1 link must have complementary settings in this field. Otherwise, the D-channel cannot function. For example, if the Avaya S8*XXX* server at one end of the link is administered as network, the other end must be administered as user. Available only when this DS1 link is providing an ISDN-PRI connection in a private network.

Related Links

Connect on page 211

Private network applications in the U.S.

Valid Entry	Usage
network	The server overrides the other end when glare occurs, and when connecting the server to a host computer.
user	The server releases the contested circuit and looks for another when glare occurs, and when connecting the server to a public network.

Private network applications outside the U.S.

Valid Entry	Usage
peer-master	The switch overrides the other end when glare occurs.
peer-slave	The switch releases the contested circuit and looks for another when glare occurs.

Interface Companding

The companding algorithm expected by the system at the far end.

Valid Entry	Usage
a-law	Algorithm expected at the far-end for E1 service.
mu-law	Algorithm expected at the far-end for T1 service.

Interworking Message

Determines what message Communication Manager sends when an incoming ISDN trunk call is routed over a non-ISDN trunk group.

Valid Entry	Usage
PROGress	Requests the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk. Normally-selected value.
ALERTing	Causes the public network in many countries to play ringback tone to the caller. This value is used only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

ITN-C7 Long Timers

Controls the T302 and T303 timers.

Available only if the **Signaling Mode** is isdn-pri.

Valid Entry	Usage
y	Increases the length of the long timers.
n	Uses the default long timers.

Related Links

Signaling Mode on page 223

Line Coding

Selects the type of line coding used on this facility. The setting in this field must match the setting on the far-end of the link, or there must be an intervening CSU to convert the line coding protocols. Voice calls work even if line coding is different match, but a single data call brings down the DS1 facility.

Caution:

If you change this field, you must busy out the DS1 circuit pack. You must also change the administration for: Route-Pattern, Access Endpoint, PRI Endpoint, Signaling-Group, and Trunk-Group.

Note:

When the DS1 circuit pack is used for ISDN service, the ISDN D-channel data is inverted forami-basic or ami-zcs is used and kept upright forb8zs or hdb3 is used.

Valid Entry	Usage
b8zs	Bipolar eight zero substitution. For T1 facilities that support voice or data traffic. Provides a 64K clear channel.
ami-zcs	Alternate mark inversion - zero code suppression. For T1 facilities that carry voice traffic. Do not use this option for digital-data applications. Use if this facility is going to be upgraded to ISDN.
ami-basic	Alternate mark inversion-basic. For unrestricted E1 facilities.
hdb3	High density bipolar 3. For restricted E1 facilities.
cmi	Coded mark inversion. Used in Japan as the only type of line coding used with the Japanese 2 Mbit trunk circuit pack.

Line Compensation

The appropriate entry in this field varies with the type of cable used. Contact your network service provider for the correct setting.

Cable lengths for a DSX-1 cross-connect

The following valid entries are for the different lengths of 22-gauge ABAM cable terminated on a DSX-1 cross-connect.

Valid Entry	Usage
1	Length: 000 – 133 (ft), 000 – 40.5 (m)
2	Length: 133 – 266 (ft), 40.5 – 81.0 (m)
3	Length: 266 – 399 (ft), 81.0 – 122 (m)
4	Length: 399 – 533 (ft), 122 – 163 (m)
5	Length: 533 – 655 (ft), 163 – 200 (m)

Cable lengths for a DS1 interface

The following valid entries are for the different lengths of 22-gauge ABAM cable directly connected to DS1 interfaces.

Valid Entry	Usage
1	Length: 0000 – 0266 (ft), 000 – 081(m)
2	Length: 0266 – 0532 (ft), 081 – 162 (m)
3	Length: 0532 – 0798 (ft), 162 – 243 (m)
4	Length: 0798 – 1066 (ft), 243 – 325 (m)
5	Length: 1066 – 1310 (ft), 325 – 400 (m)

Location

The port address.

MMI Cabling Board

Available only with the Multimedia Call Handling feature.

Valid Entry	Usage
xxxxx	Location of the multimedia interface circuit pack that is connected to the Expansion Services Module (ESM). The location can be a the cabinet, carrier, or slot address.

Related Links

Multimedia Call Handling (Basic) on page 803
Multimedia Call Handling (Enhanced) on page 803

MMI Interface

This field displays the type of multimedia (MMI) interface. The system displays this field only when you enable the **MMCH** and you enter a value for the **MMI Cabling Board**.

Related Links

Multimedia Call Handling (Basic) on page 803
Multimedia Call Handling (Enhanced) on page 803
MMI Cabling Board on page 217

Name

Assigns a significant, descriptive name to the DS1 link. Use the vendor's circuit ID for the link in this field because that information helps troubleshoot problems with the link. This field can also be used to indicate the function or the destination of this DS1 facility. Accepts up to 15 characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Peer Protocol

Administers the peer level protocol that operates in a private network. Available only if **Interface** is peer-master or peer-slave.

Valid Entry	Usage
Q-SIG	This implements QSIG Network Basic Call. Available only if Basic Call Setup is enabled for the system.
TTC	For private networking. Requires a Digital Trunk (Japan 2 MB TTC) (TN2242) circuit pack.

Related Links

Basic Call Setup on page 813
Interface on page 214

Protocol Version

Available only when:

- The **Signaling Mode** is isdn-pri and the **Connect** type is network.
- The **Signaling Mode** is isdn-pri, the **Connect** typeis pbx, and the **Interface** type is user or network.

Valid Entry	Usage
a, b, c, d	Selects the protocol that matches the network service provider's protocol in countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service. Contact the network service provider to verify that the protocols match.

Warning:

The AT&T Switched Network Protocol prohibits restricted displays of connected numbers. Display problems occur if you administer the 1a country-protocol/ protocol-version combination on the DS1 screen and administer the ISDN-PRI Trunk Group to restrict sending the connected number.

Related Links

Signaling Mode on page 223
Connect on page 211
Interface on page 214

Public network signaling administration for ISDN-PRI Layer 3

The following table describes the Communication Manager public-network access connections for ISDN-PRI Layer 3.

Admin Value	Country	Protocol Supported	B-channel mtce msg
1-a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1-b	United States	Telcordia Technologies TR 1268; NIUF.302; ANSI T1.607	Restart
1-c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United states	Telecordia Technologies SR-4287	Service
2-a	Austrailia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2-b	Austrailia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/ DPNSS through external converter)	Restart
10-b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	None

Table continues…

Admin Value	Country	Protocol Supported	B-channel mtce msg
11	Spain	Telefonica ISDN Specification	Restart
12-a	France	VN4 (French National PRI)	None
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13-a	Germany	FTZ 1 TR 6 (German National PRI)	None
13–b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI-a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI-b		ETS 300 102	None

Received Digital Metering Pulse Maximum (ms)

Available only when the **Signal Mode** is cas (Channel Associated Signaling), the **Interconnect** type is co or pbx, and the **Country Protocol** is administered for a protocol that uses periodic pulse metering (PPM).

Valid Entry	Usage
20 to 1000 ms in increments of 10ms.	This value must be greater than the **Received Digital Metering Pulse Minimum** value and match the value used by the network services provider.

Related Links

Signaling Mode on page 223
Country Protocol on page 211
Received Digital Metering Pulse Minimum (ms) on page 221

Interconnect on page 214

Received Digital Metering Pulse Minimum (ms)

Available only when the **Signal Mode** is cas (Channel Associated Signaling), the **Interconnect** type is co or pbx, and the **Country Protocol** is administered for a protocol that uses periodic pulse metering (PPM).

Valid Entry	Usage
20 to 1000 ms in increments of 10ms	This value must be greater than the **Received Digital Metering Pulse Maximum** value and match the value used by the network services provider.

Related Links

Received Digital Metering Pulse Maximum (ms) on page 220
Signaling Mode on page 223
Country Protocol on page 211
Interconnect on page 214

Received Digital Metering Pulse Value

Available only when the **Signal Mode** is cas (Channel Associated Signaling), the **Country Protocol** is 21, and the **Interconnect** type is co or pbx.

Valid Entry	Usage
0, 1	This value must match the value used by the network services provider.

Related Links

Signaling Mode on page 223
Country Protocol on page 211
Interconnect on page 214

Incoming digital PPM

The following table provides the incoming digital PPM signaling default per country protocol code.

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
0	null	NA	NA	NA
1	U.S.	NA	NA	NA
2	Australia	80	180	0
3	Japan	NA	NA	NA
4	Italy	120	150	1

Table continues…

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
5	Netherlands	90	160	0
6	Singapore	NA	NA	NA
7	Mexico	20	180	1
8	Belgium	20	180	1
9	Saudi Arabia	NA	NA	NA
10	UK	NA	NA	NA
11	Spain	20	220	0
12	France	NA	NA	NA
13	Germany	NA	NA	NA
14	Czech Republic	20	420	1
15	Russia CIS	NA	NA	NA
16	Argentina	10	180	1
17	Greece	100	180	1
18	China	NA	NA	NA
19	Hong Kong	NA	NA	NA
20	Thailand	20	180	1
21	Macedonia Croatia	120 20	180 80	1 1
22	Poland	100	150	0
23	Brazil	NA	NA	NA
24	Nordic	NA	NA	NA
25	South Africa	160	240	0, 1

Side

Use this field to control how Communication Manager must resolve glare at layer 3 over an ISDN-PRI link in QSIG private networks. The system displays this field only if the **Interface** type is peer-master or peer-slave.

Caution:

You must correctly pair with the administration of the far-end server. If the far-end server is administered as the `b` side, you must set the **Side** field to `a`, regardless of whether the layer 2 is `peer-master` or `peer-slave`, and vice versa.

Valid entry	Usage
a	The **Interface** is peer-master. Communication Manager overrides the far-end when glare occurs.

Table continues…

Valid entry	Usage
b	The **Interface** is peer-slave. Communication Manager releases the contested circuit and looks for another when glare occurs.

Related Links

Interface on page 214

Signaling Mode

Selects the signaling method used for the DS1 link. This mode must match the method used by the network services provider.

Valid Entry	Usage
CAS	Channel Associated Signaling. Out-of band signaling with E1 service. This setting yields 30 64-kbps B-channels for voice or data transmission. Channel 0 is used for framing while channel 16 carries signaling. Used for Enterprise Mobility User (EMU)/ EC500 administration.
robbed-bit	In-band signaling with T1 service. This setting yields 24 56-kbps B-channels for voice transmission.
isdn-pri	Either T1 or E1 ISDN service. This setting supports both Facility Associated Signaling and Non-Facility Associated Signaling.
isdn-ext	Either T1 or E1 ISDN service. This setting supports only Non-Facility Associated Signaling. **Note:** NFAS is primarily a feature for ISDN-T1 connections offered by service providers in North America and Hong Kong. However, it can also be used on private-network connections, and in that context it is possible to set up NFAS using ISDN-E1 interfaces.
common-chan	Out-of-band signaling with T1 service. This setting yields 23 64-kbps B-channels for voice or data transmission. Channel 24 is used for signaling.

T303 Timer (sec)

Available only if the **Group Type** is isdn-pri.

Valid Entry	Usage
2 to 10	The number of seconds the system waits for a response from the far end before invoking Look Ahead Routing.

Related Links

Interface on page 214
Group Type on page 482

MAINTENANCE PARAMETERS

Alarm When PRI Endpoint Detached

Enables or disables an alarm when the DS1 board detects a loss of signal. Used for DS1 circuit packs connected to Roll-About Video equipment.

Available only when the **Connect** type is line-side.

Related Links

Connect on page 211

Block Progress Indicator

Blocks sending the progress indicator in the SETUP message.

Available only if the **Country Protocol** is set to 1 and the **Protocol Version** is set to b.

Valid Entry	Usage
y	Prevents the progress indicator from being sent in the SETUP message.
n	Supports sending the progress indicator.

Related Links

Country Protocol on page 211
Protocol Version on page 218

EC Configuration

The set of parameters used when cancelling echo. This information is stored in firmware on the UDS1 circuit pack.

Available only if echo cancellation is enabled for the DS1 circuit pack.

Valid Entry	Usage
1 5 to 15	Provides the most rapid adaptation in detecting and correcting echo at the beginning of a call, regardless of the loudness of the talker's voice. For very loud talkers and severe echo, the far-end talker's speech is heard as clipped when both parties talk at the same time.
2	Provides slightly slower adaptation to echo. Use if speech is often clipped when both parties talk at the same time.
3	Provides slightly slower adaptation to echo but might result in a 2 or 3 second fade on strong echo for quiet talkers. Completely removes speech clipping.
4	Used in cases of extreme echo, excessive clipping or breakup of speech. May result in slight echo or background noise.

Table continues…

Valid Entry	Usage
	Note: For the MM710, the values 1 and 4 are reversed. That is, 1 for the MM710 is the same as 4 for the TN464HP/ TN2464CP, and 4 for the MM710 is the same as 1 for the TN464HP/ TN2464CP.

Related Links

Echo Cancellation on page 225

EC Direction

This field indicates the direction of the echo that is cancelled. The system displays this field only if **Echo Cancellation** is enabled for the DS1 circuit pack.

Valid entry	Usage
inward	Cancels echo energy coming back into Communication Manager. Energy from an outgoing call is reflected from an external reflection point. The party inside Communication Manager hears the echo.
outward	Cancels echo energy going outside Communication Manager. Energy from an incoming call is reflected from an internal reflection point. The party outside Communication Manager hears the echo.

Related Links

Echo Cancellation on page 225

Echo Cancellation

Enables or disables echo cancellation on the Universal DS-1 circuit pack.

Available only if DS1 echo cancellation is enabled on the system.

Related Links

DS1 Echo Cancellation on page 798

Near-end CSU

Available only when the DS1 circuit pack is a TN767D or TN464E or later suffix model, with a **Bit Rate** of 1.544 and a **Country Protocol** of 1 (U.S.).

Valid Entry	Usage
other	No channel service unit is attached to the DS1 facility or the CSU is an external unit.
integrated	A 120A CSU module is attached to the DS1 board. This integrated channel service unit (ICSU) can accept software-administrable option downlinks. In other words, the ICSU can respond to test codes from technician's equipment and report its status. Requires administration of **Integrated CSU Options**.

Related Links

Bit Rate on page 210

Country Protocol on page 211
INTEGRATED CSU OPTIONS on page 228

Slip Detection

Slips are synchronization errors that slow digital transmissions and can cause data loss. The server maintains a slip-count record for each DS1 interface to detect errors and evaluate their severity (the type of alarm). If as many as 50 percent of those spans administered for slip detection are experiencing slips (with respect to the primary), then a decision is made to switch to the secondary.

Caution:

Always enable slip detection for DS1 circuit packs that serve as primary or secondary synchronization references.

Valid Entry	Usage
y	Maintenance software measures the slip-rate of this circuit pack and determines if it is excessive. Typically, for DS1 spans used for data applications and for spans used as synchronization references. This excludes all T1-spans connecting channel banks, unless the channel bank is externally timed. This entry enables switching between the primary and secondary synchronization references and an internal high-accuracy clock.
n	For DMI-BOS links or when testing is optional. Typically used for DS1 spans that are used exclusively for voice and that do not serve as the primary or secondary synchronization source.

DS1 circuit pack: page 2

Caution:

Do not change fields on this page without assistance from Avaya or from the network service provider.

For those circuit packs that support it, this page is available only if the **Framing Mode** is esf or the **Near-end CSU** type is integrated.

Related Links

Framing Mode on page 213
Near-end CSU on page 225

CPE LOOPBACK JACK OPTIONS

Supply CPE Loopback Jack Power

Enables or disables the DS1 board's ability to supply power to the equipment during loopback testing if a Customer Premise Equipment (CPE) Loopback Jack is installed.

ESF DATA LINK OPTIONS

Far-end CSU Address

Available only if the **Framing Mode** is esf.

Valid Entry	Usage
a, b	Administers the transmit direction address used for the **`ESF data link`** command with both integrated and external channel service units (CSU).

Related Links

Framing Mode on page 213

Network Management Protocol

Available only if the **Framing Mode** is esf.

Valid Entry	Usage
tabs	The data link can be remotely examined. Used only with circuit packs that have an integrated channel service unit (CSU).

Related Links

Framing Mode on page 213

Send ANSI-T1.403 One-Second Performance Reports

Enables or disables sending error reports from the DS1 circuit pack to the far-end server or switch. These reports are useful for network management, and are sent at 1-second intervals when enabled.

Available only if the **Framing Mode** is esf. It is used only with circuit packs that have an integrated channel service unit (CSU).

Go to the Avaya Support website at http://support.avaya.comhttp://support.avaya.com/ for current documentation, product notices, knowledge articles on how to use these reports.

Related Links

Framing Mode on page 213

INTEGRATED CSU OPTIONS

Receive ALBO (Receive Automatic Line Build-Out)

Increases the strength of incoming signals by a fixed amount to compensate for line losses. To set correctly, measure the signal loss on this specific facility.

Valid Entry	Usage
26db	Used for most applications.
36db	Used for networks that differ from public telephone network standards, such as campus networks.

Transmit LBO (Transmit Line Build-Out)

Reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your Communication Manager server (measured by cable length from the smart jack) and the nearest repeater. Where another server/switch is at the end of the circuit, as in campus environments, use the cable length between the two switches to select the appropriate setting from the table below.

Available only with an integrated near-end CSU.

Valid Entry	Usage
0db	For distances of 2,001 – 3,000 feet
-7.5db	For distances of 1,001 – 2,000
-15db	For distances of 0 – 1,000 feet
-22.5db	For mid-span repeaters

Related Links

Near-end CSU on page 225

Upon DTE LOS

Tells Communication Manager what to do if the outgoing signal from the DS1 circuit pack, or Data Terminal Equipment (DTE), to the network is lost.

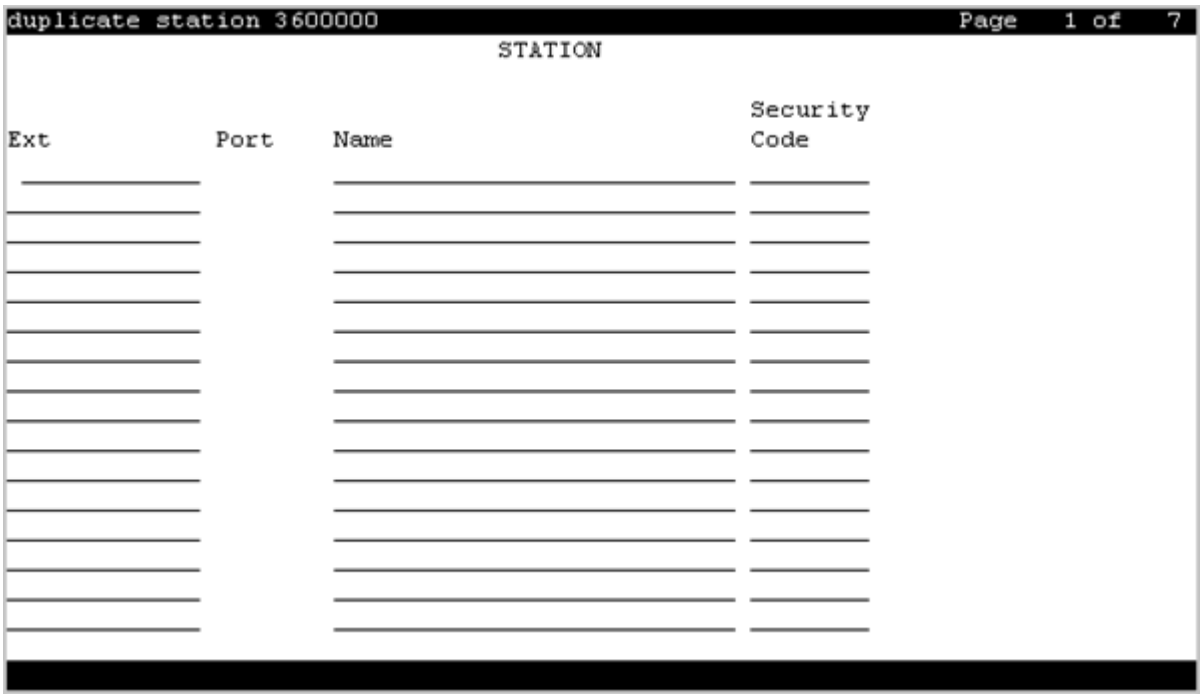
Valid Entry	Usage
loopback	Returns the network signal to the network. This prevents any alarms from being generated at the far-end.
ais	Alarm Indicator Signal. Sends an unframed all-ones signal (the AIS or Blue Alarm) to the far-end server or switch. This option alerts the network service provider to the problem immediately and aids in troubleshooting.

Chapter 44: Duplicate Station

Adds telephones by copying the information from an existing telephone and modifying this information for each new telephone. For example, configure one telephone as a template for an entire work group, and then duplicate the template station to add all the other extensions in the group. Only telephones of the same model can be duplicated. All the feature settings from the template telephone are copied to the new telephones.

Example command: `duplicate station n`, where *n* is the extension number.

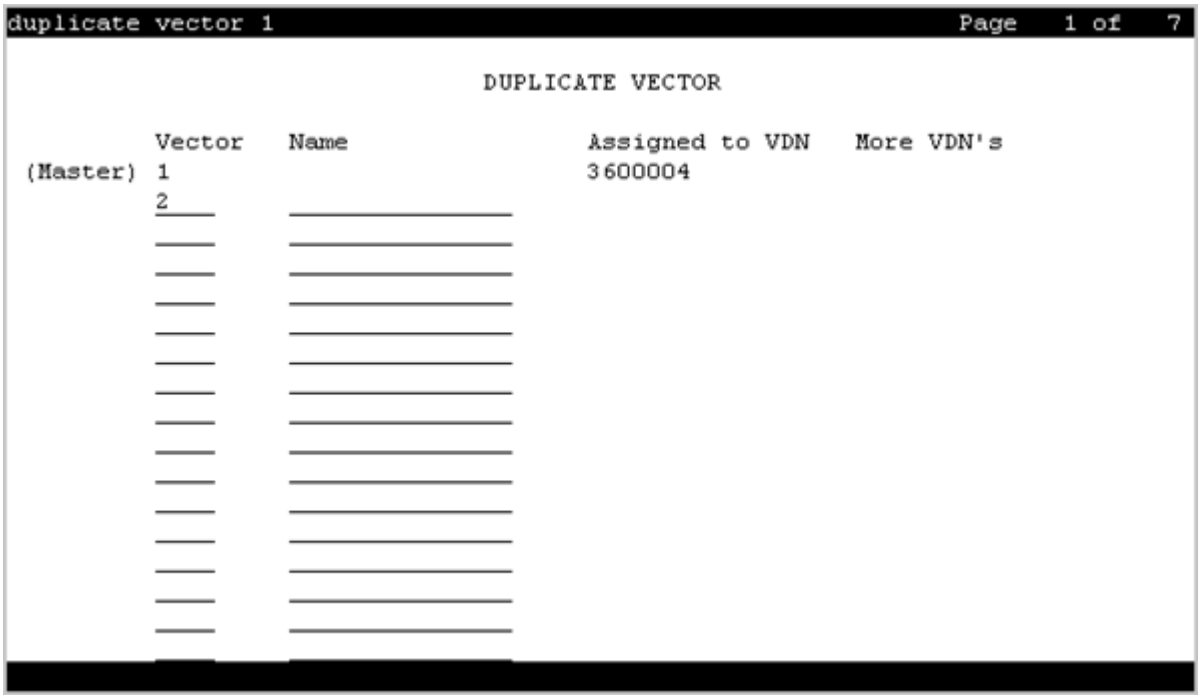
For field descriptions, see Station on page 701.



Chapter 45: Duplicate Vector

Duplicate vectors from an existing vector and edit the duplicate vectors to create vectors that are similar to the existing vector. You can use this functionality to configure one vector as a template that can be reused when creating similar vectors.

Example command: `duplicate vector n`, where *n* is the master vector.



Assigned VDN

The first assigned VDN if a VDN was assigned to the master vector.

Comments on this document? infodev@avaya.com

More VDN’s

Displays * if there is more than one VDN assigned to the same vector. For example, if 5555 displays in **VDN Assigned** and * displays in **More?** , this means that the master vector selected is already assigned to VDN 5555 as well as to other VDNs.

Name

The first row displays the vector name for the master vector if assigned. The following lines define the vector names for the duplications. The use of vector names is optional.

Vector

The first row displays the vector number for the master vector. The following lines define the vector numbers for the duplications. Any unassigned vector number is valid.

Chapter 46: Enable File Transfer

Enables SFTP on TN799BP (C-LAN) and VAL circuit packs.

Example command: `enable filexfer`

Login

The login ID. Accepts from three to six alphanumeric characters.

Password

Seven to 11 characters used as a password. The password must contain at least one number.

Secure

Enables or disables SFTP instead of FTP or TFTP. If the circuit pack does not support a secure session, no session is enabled.

Chapter 47: Enable Session

Enables Secure SHell (SSH) instead of Telnet.

Example command: `enable session`

Login

The login ID. Accepts from three to six alphanumeric characters.

Password

Seven to 11 characters used as a password. The password must contain at least one number.

Secure

Enables or disables SSH instead of Telnet.

Time to Login

Available only if the board is a TN2302.

Valid Entry	Usage
0 to 255	The number of minutes allowed for login before the session times out.

Chapter 48: Extended Pick-Up Group

Organizes pickup groups into extended pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group. With Extended Pick-Up Group, users can answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Example command: `change extended-pickup-group` *n*, where *n* is the group number.

Extended Group Number

The number associated with a collection of pickup groups.

Pickup Group Number

Valid Entry	Usage
1 to 5000 blank	The number for call pickup groups that can answer calls in the extended pickup group.

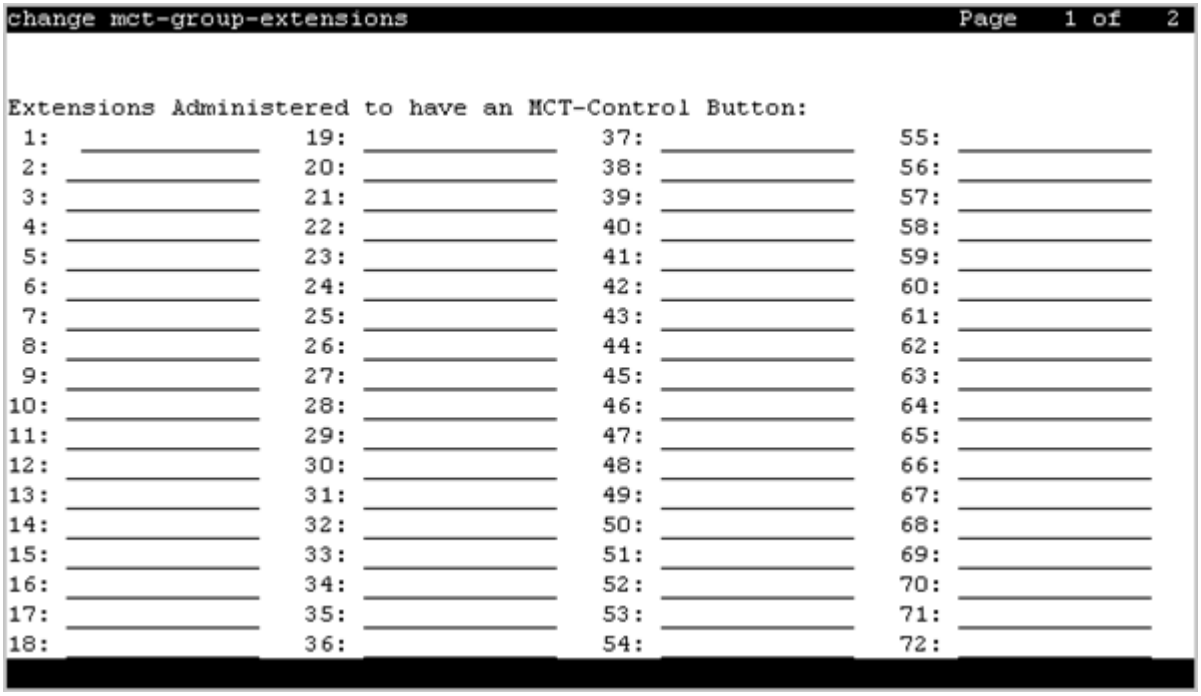
Pickup Number

The pickup number assigned to the pickup group. Users dial the pickup number after the feature access code (FAC) to pick up calls in their extended pickup group.

Chapter 49: Extensions administered to have an MCT-Control button

Lists the extensions that can take control of a Malicious Call Trace (MCT) request. To give a user the ability to take control of such requests, you musr add their extension to this list and assign them a **mct-control** feature button.

Example command: `display mct-group-extensions`



1 to 100

The extension of a telephone or attendant console that can take control of a Malicious Call Trace (MCT). A **mct-control** button must be assigned to the extension station or attendant console.

Chapter 50: Extensions to Call Which Activate Features by Name

Assigns a dialed extension to a feature within Communication Manager. This extension is called a feature name extension (FNE). The FNE mapping must be administered and all extensions must fit the dial plan. When a user calls the extension, Communication Manager activates the feature. For more information on the extensions that are paired with Feature Access Codes (FACs), see Chapter 51. For more information on the extensions that are not paired with Feature Access Codes (FACs), see Avaya Extension to Cellular User Guide, 210-100-700.

The Transfer to Voice Mail FNE is used when a user is active on a call and wants to transfer the other party to voice mail, or to the principal's voice mail, if this is a covered call. This FNE can also be used when a user goes off-hook for the first time and dials the Transfer to Voice Mail FNE to be connected to the voice mail administered in his coverage path. This is identical to dialing a Transfer to Voice Mail feature access code (FAC).

Example command: `change off-pbx-telephone feature-name-extensions`

Extension

Any valid and assigned extension number. Users call it from Cellular telephones to access Communication Manager features. Dialing the extension from their Cellular telephone is equivalent to dialing the FAC administered for the associated feature, or activating the associated feature.

Chapter 51: Failover Group Domain Mapping

A Failover Group is a group of two active Session Manager instances that are interconnected to ensure high availability of Session Manager services. Route pattern and Failover Group domain mapping are used for call preservation during a network outage. You can enter up to 9 domains in the Failover Group Domain Mapping table.

For more information on call preservation administration, see *Call Preservation Feature Description and Administration Guide*.

Example command: `display failover-grp-domain-map`

Domain

Valid Entry	Usage
A fully qualified domain name	You must type the same primary and the inverse failover group domain names, which you have already set on the Session Manager Failover Group screen. For example, if you have set the Failover Group Domain names on the Session Manager Failover Group screen as SM1-SM2.company.com (primary domain name) and SM1-SM2-inverse. company.com (inverse domain name), must set the Failover Group Domain names on the Communication Manager Failover Group screen as the same. The maximum length of the field is 64 characters. **Note:** If you enter a domain name, specify **Route Pattern**. If you set the **Domain** field to blank, set **Route Pattern** to blank.
blank	This is the default.

Related Links

Route Pattern on page 238

Route Pattern

Valid Entry	Usage
1–2000	If you are using the S8300 server, you can enter 1999 possible route patterns. If you are using the S8800, S8510, HP DL360 G7, or Dell R610 servers, you can enter 1999 possible route patterns. You must set the same route pattern in the order you defined on the Session Manager Failover Group screen. For example, if you have set the SM2 as first Session Manager and SM1 as secondary Session Manager in the route pattern list, on the Communication Manager Failover Group screen, the route pattern must have the trunk group that is associated to SM2 listed first and the trunk for SM1 listed second. You must set the **Group Type** field on the Trunk Group and the Signaling Group screens to sip and set the **Peer Server** field on the Signaling Group screen to SM for the referenced route number. **Note:** If you enter Route Pattern, do not set the **Domain** field to blank.
blank	This is the default.

Related Links

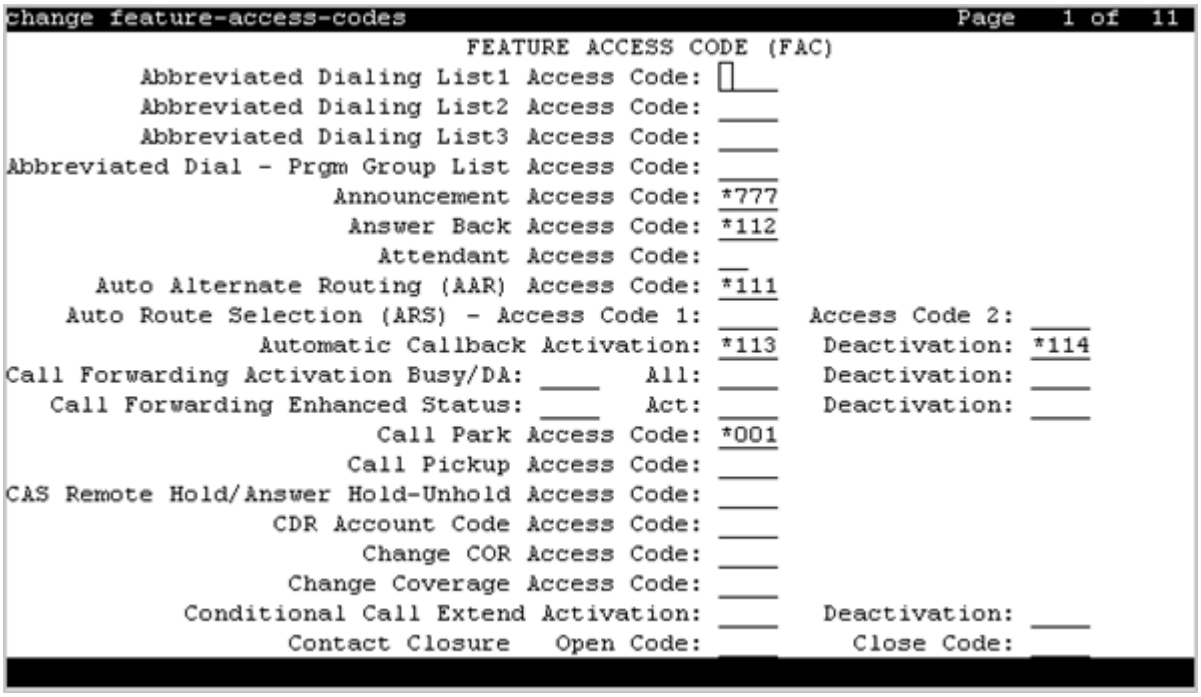
Domain on page 237
Group Type on page 675
Group Type on page 482

Chapter 52: Feature Access Code (FAC)

Assigns feature access codes (FACs) that, when dialed, activate or cancel system features.

Example command: `change feature-access-codes`

Feature Access Code: page 1



Abbreviated Dialing List1 Access Code

A feature access code (FAC) used to access AD list 1.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dialing List2 Access Code

A feature access code (FAC) used to access AD list 2.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dialing List3 Access Code

A feature access code (FAC) used to access AD list 3.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dial - Prgm Group List Access Code

FAC used to enter a group list from a telephone. The user's extension must be administered with permission to program the group list.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Program Ext on page 28

Announcement Access Code

FAC used to record announcements.

This value must conform to the FACs or dial access codes defined by the dial plan.

Answer Back Access Code

FAC used to retrieve parked calls. If no one answers the call before a system-wide expiration interval expires, the system redirects the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto Route Selection (ARS) Access Code 2

Additional FAC used to access ARS.

This value must conform to the FACs or dial access codes defined by the dial plan.

Attendant Access Code

FAC used to call the attendant. While only one attendant can be administered for the dial plan, more than one attendant FAC can be administered in a single distributed network. Attendant access numbers can start with any number from 0 to 9 and contain one or two digits. Available only if an attendant call type is not administered for the dial plan.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Call Type on page 198

Auto Alternate Routing (AAR) Access Code

FAC used to access AAR. AAR routes calls to a different route than the first-choice route when facilities are unavailable.

This value must conform to the FACs or dial access codes defined by the dial plan.

Automatic Callback Activation/Deactivation

FAC used to activate or cancel Automatic Callback. Automatic Callback enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both originating and receiving parties when the receiving party becomes available.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto Route Selection (ARS) Access Code 1

FAC used to access ARS. The system can automatically choose the least-expensive way to send a toll call. You can have one ARS access code for local and one for long distance, and route accordingly.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Activation Busy/DA

FAC used to forward calls to an administered number if the user is busy or does not answer.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Enhanced Activation/Deactivation

FAC numbers used to activate and deactivate Enhanced Call Forwarding. Enhanced Call Forwarding forwards incoming calls to different destinations depending on whether they are from internal or external sources. The FACs for activation and deactivation must be administered together.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Enhanced Status

FAC used to display the status of Enhanced Call Forwarding.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Park Access Code

FAC used to park an active call that can then be retrieved from a different station using the answer back access code. The call park access code cannot have the same first digit as another feature access code that is longer in length.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Pickup Access Code

FAC used to answer a call directed to a pickup group.

This value must conform to the FACs or dial access codes defined by the dial plan.

CAS Remote Hold/Answer Hold-Unhold Access Code

FAC used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote server running Communication Manager. This FAC can also be used by an analog station. Flashing the switch-hook for the proper interval (between 200 and 1000 ms) while talking on an existing call causes the existing call to be placed on soft hold, using which the analog user can dial the Answer Hold-Unhold FAC to Hard hold the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

CDR Account Code

FAC used prior to entering an account code for Call Detail Recording (CDR) purposes. CDR is a feature that uses software and hardware to record call data.

This value must conform to the FACs or dial access codes defined by the dial plan.

Change COR Access Code

Using FAC, users can change their class of restriction (COR) from a telephone. Available only if **Change COR by FAC** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Change COR by FAC on page 797

Change Coverage Access Code

FAC used to change a coverage path from a telephone or remote station. The coverage path is the order in which calls are redirected to alternate answering positions.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Security Code on page 736

Contact Closure Close Code

FAC used to close a contact closure relay. Contact closures control electrical devices remotely. Users use an FAC to activate electrical devices such as electrical door locks. If **Contact Closure Open Code** is administered, then **Contact Closure Close Code** must also be administered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Contact Closure Open Code on page 244

Contact Closure Open Code

FAC used to open a contact closure relay. Contact closures control electrical devices remotely. Users use an FAC to activate electrical devices such as electrical door locks. If **Contact Closure Close Code** is administered, then **Contact Closure Open Code** must also be administered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Contact Closure Close Code on page 243

Feature Access Code: page 2

```
change feature-access-codes                                  Page   2 of  11
                          FEATURE ACCESS CODE (FAC)
                Contact Closure  Pulse Code:

              Data Origination Access Code:
                  Data Privacy Access Code:
            Directed Call Pickup Access Code: *102
      Directed Group Call Pickup  Access Code:
   Emergency Access to Attendant Access Code:
      EC500 Self-Administration Access Codes:      ____   ____   ____
                  Enhanced EC500 Activation:      Deactivation:
         Enterprise Mobility User Activation:     Deactivation:
Extended Call Fwd Activate Busy D/A ____ All:     Deactivation:
      Extended Group Call Pickup Access Code:
             Facility Test Calls Access Code:
                        Flash Access Code:
           Group Control Restrict Activation:     Deactivation:
                 Hunt Group Busy Activation:     Deactivation:
                        ISDN Access Code:
              Last Number Dialed Access Code:
   Leave Word Calling Message Retrieval Lock:
 Leave Word Calling Message Retrieval Unlock:
```

Contact Closure Pulse Code

FAC used to pulse a contact closure relay.

This value must conform to the FACs or dial access codes defined by the dial plan.

Data Origination Access Code

FAC used to originate a data call from a voice station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Data Privacy Access Code

FAC used to isolate a data call from call waiting or other interruptions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Directed Call Pickup Access Code

The FAC used with the extension number for which the user wants to pick up a call. To pick up a call for an extension, the user dials the FAC followed by the extension number. The called extension number stops ringing and the user connects to the call for pickup.

The FAC value must conform to the FACs or dial access codes defined in the dial plan.

Directed Group Call Pickup Access Code

The FAC used to pickup a call from any pickup group.

The FAC value must conform to the FACs or dial access codes defined in the dial plan.

EC500 Self Administration Access Code

Using FAC, users can self-administer their cell phone number for the Extension to Cellular feature. Users can add or change their cell phone number through this feature access code. An administrator can still enter or change cell phone numbers. The user calls the Self Administration Access Code access code and enters their cell phone number.

This value must conform to the FACs or dial access codes defined by the dial plan.

Emergency Access To Attendant Access Code

FAC used to gain access to the attendant in an emergency. Such calls alert as emergency calls.

Available only if **Emergency Access to Attendant** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Emergency Access to Attendant on page 798

Enhanced EC500 Activation

Using FAC, users can activate Extension to Cellular remotely.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enhanced EC500 Deactivation

Using FAC, users can deactivate Extension to Cellular remotely.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enterprise Mobility User Activation

FAC used to activate the Enterprise Mobility User feature for a particular user, associating the features and permissions of their primary telephone to a telephone of the same type anywhere within the customer's enterprise.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enterprise Mobility User Deactivation

FAC used to deactivate the Enterprise Mobility User feature.

This value must conform to the FACs or dial access codes defined by the dial plan.

Extended Call Fwd Activate All

FAC used to activate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Security Code on page 736

Extended Call Fwd Activate Busy D/A

FAC used to activate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Security Code on page 736

Extended Call Fwd Deactivation

FAC used to deactivate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Security Code on page 736

Extended Group Call Pickup Access Code

FAC used to answer a call directed to another pickup group. Users must enter a valid pickup number following this field to complete the operation.

This value must conform to the FACs or dial access codes defined by the dial plan.

Facility Test Calls Access Code

FAC used to place a facility test call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Security alert:

To ensure the security of your system, leave this field blank except when actually testing trunks.

Flash Access Code

FAC used to generate trunk flash. This code ensures that the flash signal is interpreted by the local telephone company central office switch, rather than by Avaya Communication Manager.

This value must conform to the FACs or dial access codes defined by the dial plan.

Group Control Restrict Activation/Deactivation

FAC used to change the restriction level for all users with a given class of restriction. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Hunt Group Busy Activation/Deactivation

Hunt group members use the **Hunt Group Busy Activation** FAC to make the extension unavailable and the **Hunt Group Busy Deactivation** FAC to make the extension available.

The hunt group member must dial:

- a two-digit hunt group number for a hunt group supporting up to 99 extensions (small hunt group) or
- a four-digit hunt group number for a hunt group supporting up to 8000 extensions (large hunt group)

If needed, the hunt group member must add zeros before the hunt group number to ensure that the small hunt group has a two-digit number and the large hunt group has a four-digit number.

This value must conform to the FACs or dial access codes defined by the dial plan.

ISDN Access Code

FAC used to place an ISDN call without using ARS, AAR, or UDP.

This value must conform to the FACs or dial access codes defined by the dial plan.

Last Number Dialed Access Code

FAC used to redial the last number dialed from this station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Leave Word Calling Message Retrieval Lock

FAC used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. Available only if **Lock Messages** is administered for the station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Lock Messages on page 723

Leave Word Calling Message Retrieval Unlock

FAC used to unlock a telephones display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 3

```
change feature-access-codes                                  Page   3 of  11
                            FEATURE ACCESS CODE (FAC)
            Leave Word Calling Send A Message: ____
          Leave Word Calling Cancel A Message: ____
  Limit Number of Concurrent Calls Activation: ____   Deactivation: ____
              Malicious Call Trace Activation: ____   Deactivation: ____
        Meet-me Conference Access Code Change: ____
         Message Sequence Trace (MST) Disable: ____

PASTE (Display PBX data on Phone) Access Code: ____
 Personal Station Access (PSA) Associate Code: ____   Dissociate Code: ____
       Per Call CPN Blocking Code Access Code: ____
     Per Call CPN Unblocking Code Access Code: ____
                  Posted Messages Activation: ____   Deactivation: ____
                Priority Calling Access Code: ____
                         Program Access Code: ____

      Refresh Terminal Parameters Access Code: ____
             Remote Send All Calls Activation: ____   Deactivation: ____
              Self Station Display Activation: ____
                    Send All Calls Activation: *100   Deactivation: *101
        Station Firmware Download Access Code: ____
```

Leave Word Calling Cancel A Message

FAC used to cancel a leave word calling message.

This value must conform to the FACs or dial access codes defined by the dial plan.

Leave Word Calling Send A Message

Using FAC, users can send a leave word calling message that allows internal system users to leave a short pre-programmed message for other internal users.

This value must conform to the FACs or dial access codes defined by the dial plan.

Limit Number of Concurrent Calls Activation/Deactivation

Use this field to configure the feature access codes to enable or disable the Limit Number of Concurrent Calls feature. If a user dials the activation FAC, Communication Manager restricts the number of incoming calls on the station to one call at a time. If the station user is busy, the subsequent incoming calls receive a busy tone or are transferred to voice mail.

Note:

Communication Manager Release 6.2 and earlier supported the Limit Number of Concurrent Calls feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

Malicious Call Trace Activation

FAC used to activate a trace request on a malicious call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Meet-me Conference Access Code Change

Using FAC, the controlling user of a Meet-me Conference VDN can change the access code. The Meet-me Conference feature is used to set up a dial-in conference of up to six parties.

This value must conform to the FACs or dial access codes defined by the dial plan.

Message Sequence Trace (MST) Disable

Provides the ability to disable the MST traces using a Feature Access Code.

PASTE (Display PBX data on telephone) Access Code

FAC used to view call center data on display telephones. PASTE is used in conjunction with Avaya IP Agent.

This value must conform to the FACs or dial access codes defined by the dial plan.

Per Call CPN Blocking Code Access Code

Use this FAC to activate Calling Party Number (CPN) blocking of a station. When users dial the code, the calling party information is not sent to the public network.

This value must conform to the FACs or dial access codes defined by the dial plan.

Per Call CPN Unblocking Code Access Code

Use this FAC to deactivate Calling Party Number (CPN) blocking of a station. When users dial the code, the calling party information is sent to the public network.

This value must conform to the FACs or dial access codes defined by the dial plan.

Personal Station Access (PSA) Associate Code

FAC used to associate a telephone with the telephone features assigned to a users extension.

Available only if **Personal Station Access (PSA)** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Personal Station Access (PSA) on page 804

Personal Station Access (PSA) Dissociate Code

FAC used to remove the association between a physical telephone and an extension number.

Available only if **Personal Station Access (PSA)** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Personal Station Access (PSA) on page 804

Posted Messages

FAC used to access the Posted Messages feature. The Posted Messages feature provide callers with a displayed message on the telephone that states why the user is unavailable to take a call.

Available only if the Posted Messages feature is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Posted Messages on page 804

Priority Calling Access Code

FAC used to enable priority calling, a special type of call alerting between internal telephone users, including the attendant. The called party hears a distinctive ringing when the calling party uses Priority Calling.

This value must conform to the FACs or dial access codes defined by the dial plan.

Program Access Code

FAC used to program abbreviated dial buttons on an individual telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

Refresh Terminal Parameters Access Code

Feature Access Code (FAC) is used to request a refresh of the terminal parameters on a telephone that supports downloadable parameters. This FAC is used after a DCP telephone is installed or replaced to ensure that all the terminal parameters, including button labels, are sent to the telephone.

Remote Send All Calls Activation/Deactivation

FAC used to activate or deactivate the Send All Calls feature. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Self Station Display Activation

FAC used on a digital station to display its primary extension number when the FAC is entered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Send All Calls Activation/Deactivation

FAC used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

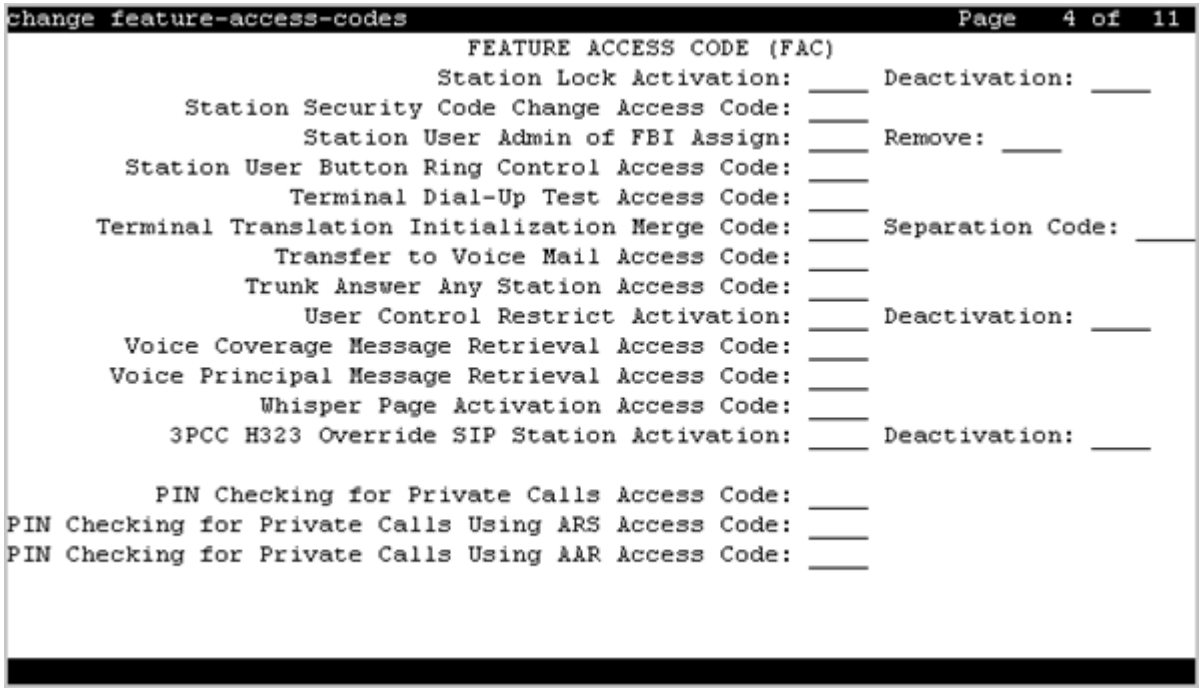
This value must conform to the FACs or dial access codes defined by the dial plan.

Station Firmware Download Access Code

FAC used for 2420/2410 DCP station firmware downloads.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 4



3PCC H323 Override SIP Station Activation/Deactivation

FAC used to activate and deactivate third party call control (3PCC) redirect actions from a SIP desktop station to a remote softphone or virtual private network H.323 telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

PIN Checking for Private Calls Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and that prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

PIN Checking for Private Calls on page 343

PIN Checking for Private Calls Using AAR Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

If a user dials and assigns this FAC, the Call Processing feature treats the call as if the user has dialed the FAC for AAR explicitly after entering the PIN code.

Related Links

PIN Checking for Private Calls on page 343

PIN Checking for Private Calls Using ARS Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

If a user dials and assigns this FAC, the Call Processing feature treats this call as if the user has dialed the FAC for ARS explicitly after entering the PIN code.

Related Links

PIN Checking for Private Calls on page 343

Station Lock Activation/Deactivation

FAC used to activate or deactivate Station Lock. The Station Lock feature locks a telephone to prevent others from placing outgoing calls from the telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station Security Code Change Access Code

FAC used to change a Station Security Code (SSC). The SSC feature is used to deny other users access to the functions that are associated with the station. Each station user can change their own SSC if they know the current settings for the station. The SSC must be administered before the user can change it using this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station User Admin of FBI Assign

FAC used to activate or deactivate Facility Busy Indicators that provide visual indicators of the busy or idle status of any particular trunk group, hunt group member, or station user.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station User Button Ring Control Access Code

FAC used to control the ring behavior for each line appearance and bridged appearance from the station. Users can have their telephones ring either silently or audibly.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Dial-Up Test Access Code

FAC used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the server running Avaya Communication Manager. The Terminal Dial-Up test ensures that the terminal and each of its buttons can communicate with the server. This test is initiated by a user entering this feature access code. This test is mostly for use by terminal service personnel, but can be used by any station user.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Translation Initialization Merge Code

FAC used to install, or merge, a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must already have been activated or the station administered without hardware, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Translation Initialization Separation Code

FAC used to remove, or separate, a station from a location without losing any of its feature settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Transfer to Voice Mail Access Code

FAC used to activate coverage to transfer the caller to the original call recipient's voice mail where the caller can leave a message. This FAC cannot have the same first digit as another FAC that is longer in length.

This value must conform to the FACs or dial access codes defined by the dial plan.

Trunk Answer Any Station Access Code

The FAC used to answer calls alerting on night bells.

This value must conform to the FACs or dial access codes defined by the dial plan.

User Control Restrict Activation/Deactivation

FAC used to activate and deactivate specific restrictions for an individual user or an attendant. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Coverage Message Retrieval Access Code

FAC used to retrieve voice messages for another user used as a coverage point, using a digital display module.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Principal Message Retrieval Access Code

FAC used by a user to retrieve their own voice messages for another user using a digital display module.

This value must conform to the FACs or dial access codes defined by the dial plan.

Whisper Page Activation Access Code

FAC used to place a page to another user's telephone when active on a call. Only the paged user hears the page, not the other parties on the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 5

```
change feature-access-codes                                  Page   5 of  11
                              FEATURE ACCESS CODE (FAC)
                                Call Center Features
 AGENT WORK MODES
                          After Call Work Access Code:
                                  Assist Access Code:
                                Auto-In Access Code:
                               Aux Work Access Code:
                                  Login Access Code:
                                 Logout Access Code:
                               Manual-in Access Code:
 SERVICE OBSERVING
           Service Observing Listen Only Access Code:
           Service Observing Listen/Talk Access Code:
               Service Observing No Talk Access Code:
  Service Observing Next Call Listen Only Access Code:
Service Observing by Location Listen Only Access Code:
Service Observing by Location Listen/Talk Access Code:

 AACC CONFERENCE MODES
                    Restrict First Consult Activation: #31  Deactivation: #32
                   Restrict Second Consult Activation: #33  Deactivation: #34
```

AACC Conference Modes

Avaya Aura® Contact Center uses the Agent Mode feature access codes to restrict the call transfers or conferences that an agent can initiate. In the Restrict First Consult mode, an agent can initiate only one conference or one transfer sequence at a time. When an agent initiates a call transfer or a conference, the system matches the held call appearance and the initiated call appearance. Before initiating a new conference or a transfer sequence, the agent must either complete the transfer with one of the two appearances or drop one of the calls.

Note:

The AACC Conference modes are not supported on ISDN-BRI telephones.

Comments on this document? infodev@avaya.com

Restrict First Consult Activation/Deactivation

Use this field to enable the Restrict Call Joining feature. The Restrict Call Joining feature is supported only on Avaya Aura® Contact Center. With this feature, Communication Manager restricts the agents from initiating a transfer or a conference operation.

Restrict Second Consult Activation/Deactivation

Use this field to enable the Restrict Second Agent Consult feature. This feature is supported only on Avaya Aura® Contact Center. With this feature, Communication Manager allows the agents to use only one consult operation at a time. The consult operations include transfer and conference.

Note:

If you set the **Restrict Second Call Consult** field to n, after you dial the **Restrict Second Consult** feature access code, the feature access code overrides the setting of the **Restrict Second Call Consult** field on the COR group of the station.

Add Agent Skill Access Code

FAC dialed by an agent to add a skill to their current skill set.

This value must conform to the FACs or dial access codes defined by the dial plan.

After Call Work Access Code

FAC dialed by an agent when the agent performs work-related Automatic Call Distribution (ACD) activities.

This value must conform to the FACs or dial access codes defined by the dial plan.

Assist Access Code

FAC dialed by an agent to request assistance from the split supervisor. The split supervisor is someone working in a call center that uses various Communication Manager and Call Management System features to check split and agent performance and to provide assistance if necessary.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto-In Access Code

FAC dialed when an agent is ready to process another call as soon as the current call is completed.

This value must conform to the FACs or dial access codes defined by the dial plan.

Aux Work Access Code

FAC used when an agent is unavailable to receive ACD, or work-related, calls. Agents use this FAC for activities such as taking a break, going to lunch, or placing an outgoing call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Login Access Code

FAC dialed by an agent to gain access to the ACD functions. This is a system-wide code for all ACD agents.

This value must conform to the FACs or dial access codes defined by the dial plan.

Logout Access Code

FAC dialed by the agent to exit ACD. This is a system-wide logout code for all ACD agents.

This value must conform to the FACs or dial access codes defined by the dial plan.

Manual-In Access Code

FAC dialed when an agent is ready to process another call manually.

This value must conform to the FACs or dial access codes defined by the dial plan.

Remote Logout of Agent Access Code

FAC typically used by a supervisor to logout an idle agent without being physically present at the agent station. The supervisor can be locally or remotely located. Available only if **Service Observing (Remote/By FAC)**, **Vectoring (Basic)**, and **Vectoring (Prompting)** are enabled on the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Service Observing (Remote/By FAC) on page 810
Vectoring (Basic) on page 810
Vectoring (Prompting) on page 811

Remove Agent Skill Access Code

FAC dialed by an agent to remove a skill from their current skill set. Available only if **Service Observing (Remote/By FAC)** is enabled on the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Service Observing (Remote/By FAC) on page 810

Service Observing Listen Only Access Code

Dial FAC for a station with Service Observing permission to listen to other agent ACD calls without being heard on the ACD call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Service Observing Listen/Talk Access Code

Dial FAC for a station with Service Observing permission to both listen and be heard on an ACD call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Service Observing Next Call Listen Only Access Code

You can assign a Feature Access Code (FAC) to the Service Observing Next Call Listen Only Access Code feature. When you dial the Service Observing Next Call Listen Only Access Code, Communication Manager waits until the current call ends that the extension to be service observed is on. Communication Manager then starts service observing at the beginning of that station's next call. You can listen to the other call, but not speak into it.

Service Observing No Talk Access Code

Dial FAC for a station with Service Observing permission to listen only. Any attempt to toggle between listening and talking using the Service Observing button is denied. Available only if **Expert Agent Selection (EAS) Enabled** is administered for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Expert Agent Selection (EAS) Enabled on page 314

Feature Access Code: page 6

```
change feature-access-codes                                  Page   6 of  11
                              FEATURE ACCESS CODE (FAC)

                                Call Center Features

FORCED LOGOUT/AUX
    Forced Agent Logout by Location Access Code: ____
       Forced Agent Logout by Skill Access Code: ____
  Forced Agent Aux Work by Location Access Code: ____
     Forced Agent Aux Work by Skill Access Code: ____

MISCELLANEOUS
                     Add Agent Skill Access Code: ____
                  Remove Agent Skill Access Code: ____
              Remote Logout of Agent Access Code: ____
```

Converse Data Return Code

FAC used to pass values between a Voice Response Unit (VRU) and Communication Manager to play an announcement, collect digits from the caller, and so on.

Vector Variable x

FAC (# can be used as the first digit), used to change the value of defined variables.

Related Links

Variables for Vectors on page 943

Feature Access Code: page 7

```
change feature-access-codes                                  Page   7 of  11
                              FEATURE ACCESS CODE (FAC)

                         Call Vectoring/Prompting Features

   Converse Data Return Code: ____

Vector Variable 1 (VV1) Code: ____
Vector Variable 2 (VV2) Code: ____
Vector Variable 3 (VV3) Code: ____
Vector Variable 4 (VV4) Code: ____
Vector Variable 5 (VV5) Code: ____
Vector Variable 6 (VV6) Code: ____
Vector Variable 7 (VV7) Code: ____
Vector Variable 8 (VV8) Code: ____
Vector Variable 9 (VV9) Code: ____
```

Automatic Wakeup Call Access Code

FAC dialed to schedule or cancel a wakeup call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Housekeeping Status (Client Room) Access Code

FAC the housekeeper dials from a client room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing.

This value must conform to the FACs or dial access codes defined by the dial plan.

Housekeeping Status (Station) Access Code

FAC the housekeeper dials to provide room status. This access code must be dialed from designated telephones.

This value must conform to the FACs or dial access codes defined by the dial plan.

Verify Wakeup Announcement Access Code

FAC dialed to verify a wakeup announcement.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Do Not Disturb Access Code

FAC dialed to enter or cancel a “do not disturb” request using voice prompting.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 8

```
change feature-access-codes                                  Page   8 of  11
                            FEATURE ACCESS CODE (FAC)
                              Hospitality Features

             Automatic Wakeup Call Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
  Housekeeping Status (Client Room) Access Code: ____
      Housekeeping Status (Station) Access Code: ____
      Housekeeping Status (Station) Access Code: ____
      Housekeeping Status (Station) Access Code: ____
      Housekeeping Status (Station) Access Code: ____
         Verify Wakeup Announcement Access Code: ____
               Voice Do Not Disturb Access Code: ____
```

Basic Mode Activation

FAC dialed to revert an Enhanced multimedia complex to a Basic multimedia complex.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enhanced Mode Activation

FAC used to convert a Basic multimedia complex to an Enhanced multimedia complex.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Call Access Code

FAC that indicates to Avaya Communication Manager that an Enhanced mode multimedia call is being made. This FAC originates a multimedia call according to system-wide default settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Default Multimedia Outgoing Trunk Parameter Selection on page 275

Multimedia Data Conference Activation

FAC that when entered from any voice station that is participating in a multimedia call, alerts Avaya Communication Manager to enable data collaboration with the other parties on the call. If this FAC is entered a second time, multimedia data conference activation is denied since it is already active. This FAC only applies to voice stations on servers equipped with ESM adjuncts.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Data Conference Deactivation

FAC that when entered from the telephone deactivates the data session and reverts to a voice and video call. This telephone supports data collaboration on a multimedia mode call. If a user enters this FAC while participating in a data collaboration multimedia call that the user did not initiate, the system denies the deactivation.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Multi-Address Access Code

FAC that supports origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2B-channels.

For example, ISDN-BRI provided by a local telephone company central office is provisioned with separate listed directory numbers for each B-channel. To make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code originates a call according to system-wide default settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related Links

Default Multimedia Outgoing Trunk Parameter Selection on page 275

Multimedia Parameter Access Code

FAC that prompts Avaya Communication Manager to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 9

```
change feature-access-codes                                  Page   9 of  11
                            FEATURE ACCESS CODE (FAC)
                               Multimedia Features

                      Basic Mode Activation: ____
                  Enhanced Mode Activation: ____
               Multimedia Call Access Code: ____
Multimedia Data Conference Activation: ____    Deactivation: ____
 Multimedia Multi-Address Access Code: ____
     Multimedia Parameter Access Code: ____
```

Precedence Calling Access Code

FAC used to access the Multiple Level Precedence and Preemption (MLPP) feature using which users can request priority processing of calls during critical situations.

This value must conform to the FACs or dial access codes defined by the dial plan.

WNDP PRECEDENCE ACCESS CODES

FAC used to determine the precedence level for a call when the Worldwide Numbering Dial Plan (WNDP) feature is active. The WNDP feature is compatible with the standard numbering system that the Defense Communications Agency (DCA) established. Different feature access codes are assigned for each PRECEDENCE level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Flash Access Code

FAC that corresponds to the Flash preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Flash Override Access Code

FAC that corresponds to the Flash Override preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Immediate Access Code

FAC that corresponds to the Immediate preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Priority Access Code

FAC that corresponds to the Priority preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Routine Access Code

FAC that corresponds to the Routine preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Destination Code Control

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Control DCC activation and deactivation by dialing the Activation code and the Deactivation code. DCC activates and deactivates the functionality systemwide.

Activation Code

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The FAC used to activate the DCC feature.

This value must conform to the FACs defined by the dial plan.

Deactivation Code

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The FAC used to deactivate the DCC feature.

This value must conform to the FACs defined by the dial plan.

Chapter 53: Feature-related system parameters

This screen implements system parameters associated with various system features.

Note:

Call Coverage and Call Forwarding parameters are located on the System Parameters Call Coverage / Call Forwarding screen.

Feature-related system parameters: page 1

```
change system-parameters features                              Page   1 of  19
                          FEATURE-RELATED SYSTEM PARAMETERS
                             Self Station Display Enabled? n
                                  Trunk-to-Trunk Transfer: none
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? no
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n



            Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```


Comments on this document? infodev@avaya.com

AAR/ARS Dial Tone Required

Enables or disables a second dial tone that tells the user that additional dialing can occur. The second dial tone is provided on a incoming tie or DID trunk call routed through AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid Entry	Usage
y	Programming is done by the station-assigned list.
n	Indicates that a Program Access code is being used to indicate which personal list is to be programmed.

ACA Long Holding Time Originating Extension

The extension number that the ACA feature uses when sending a long holding time referral call. This extension number must be different from the extension number used for **ACA Short Holding Time Originating Extension**. Available for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

ACA Referral Calls

Indicates where Automatic Circuit Assurance (ACA) referral calls generate. The Automatic Circuit Assurance (ACA) feature is used to identify possible trunk malfunctions. If a possible trunk malfunction is identified, Communication Manager sends the attendant a referral call that consists of a display message or a voice-synthesized message indicating the problem and where the problem is located. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
local	Generate on and for the local switch.
primary	Generate on the local switch for remote servers and switches as well as the local switch.
remote	Generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number administered for the dial plan. Also, **ACA** button status transmits to other servers and switches when in a DCS network.

ACA Referral Destination

The specified extension or attendant must be equipped with a display module. Available only for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
extension number	The extension on a local server running Communication Manager receives the ACA referral call.
attd	The attendant receives the ACA referral call.

ACA Remote PBX Identification

Available for remote ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
1 to 63	Identifies the switch in a DCS network that makes the referral call. The switch identified here cannot be the remote server or switch identified as **local** on the system Dial Plan.

ACA Short Holding Time Originating Extension

The extension number that the ACA feature uses when sending a short holding time referral call. This extension number must be different from the extension number used for **ACA Long Holding Time Originating Extension**.

Available for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid Entry	Usage
1 to 16	The number of rings before an automatic abbreviated or delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid Entry	Usage
2 to 9	The number of times the callback rings at the calling station before the callback is canceled. Using Automatic Callback, internal users who place a call to a busy or an unanswered internal telephone can be called back when the called telephone becomes available.

Automatic Circuit Assurance (ACA) Enabled

Enables or disables the ACA feature to measure the holding time of each trunk call to identify possible trunk malfunctions. Long duration thresholds are designed to detect trunks that have been connected beyond the limit. Short duration thresholds detect trunks that might be dropping. Requires that an **aca-halt** button is administered on the user station.

Call Park Timeout Interval (minutes)

Valid Entry	Usage
1 to 90	The number of minutes a call remains parked before it cancels. Using the Call Park, users can retrieve a call on hold from any other telephone within the system.

DID/Tie/ISDN/SIP Intercept Treatment

Valid Entry	Usage
announcement	Routes intercepted calls to a recorded announcement that indicates that the number reached is inoperative or restricted. **Note:** This intercept treatment is provided as a system option. Only one message is given to the user.
attendant	Routes intercept calls to the attendant so that the attendant knows when a problem exists.
vdn	Routes intercepted calls to a vector directory number.

Display Calling Number for Room to Room Caller ID Calls

Enables or disables displaying the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

Using Internal Automatic Answer (IAA), users can answer internal calls on digital telephones (except BRI) using a speakerphone or a headset .

Valid Entry	Usage
attd-extended	Enables IAA for only attendant-extended calls.
both	Enables IAA for station-transferred and attendant-extended calls.
none	Disables IAA for all calls.
transferred	Enables IAA for only station-transferred calls.

Music (or Silence) On Transferred Trunk Calls

Valid Entry	Usage
all	All transferred trunk calls receive music until the call is answered if the Music-on-Hold feature is available.
no	When the Trunk callers are waiting to be answered, they hear music or nothing if Music-on-Hold is not administered. When the transfer is completed, the callers then hear ringback till the call is answered.
call-wait	Trunk calls are transferred to stations that require the call to wait and hear music, if administered. All other transferred trunk calls receive ringback tone.

Music/Tone on Hold

Indicates what a caller hears while on hold. Available only if **Tenant Partitioning** is disabled for the system.

Valid Entry	Usage
music	The caller hears music while on hold.
tone	The caller hears a tone while on hold.
none	The caller hears silence while on hold.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid Entry	Usage
5 to 25	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses

Valid Entry	Usage
	on the trunk or gets intercept treatment depending on the **Outpulse Without Tone** settings administered for the system.

Port

Available only when the music on hold type is port.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks). 01 to 31	Sixth and seventh characters are the circuit number
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Protocol for Caller ID Analog Terminals

Determines the protocol or tones sent to a Caller ID telephone. The provider should have the correct protocol for each country.

Valid Entry	Usage
Bellcore	Telcordia Technologies protocol with 212 modem protocol tones. Typically used in the U.S.
V23–Bell	Telcordia Technologies protocol with V.23 modem tones. Typically used in Bahrain.

Self Station Display Enabled

Enables or disables the displaying of the primary extension associated with a digital display telephone when the **inspect** button is pressed or when a feature access code (FAC) is entered.

- When using an FAC, the display continues until the user picks up the telephone or receives an incoming call.
- When using the **inspect** button, the display continues until the user presses the **normal** or **exit** button, or until the user picks up the telephone or receives an incoming call.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. For more information, go to the Avaya Support website at http://support.avaya.com/ for current documentation, product notices, knowledge articles.

Valid Entry	Usage
all	Enables all trunk-to-trunk transfers. With this, telephone users can set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location. This value is required for SIP support.
restricted	Restricts all public trunks (CO, WATS, FX, CPE, DID, and DIOD) from being transferred.
none	Restricts all trunks (except CAS and DCS) from being transferred.

Type

Indicates whether the source for Music on Hold is an announcement extension, an audio group, or a port on a VAL board.

Available only if music is administered as the treatment for calls placed on hold.

Valid Entry	Usage
ext	Extension. Requires entry of the corresponding extension number of the announcement or audio source.
group	Analog group. Requires entry of the corresponding Music-on-Hold analog group number.
port	Analog/aux-trunk. Requires entry of the corresponding location of the Music-on-Hold analog or aux-trunk source. **Note:** A source identifier (extension number, audio group number, or port number) must be administered with the source type.

Feature-related system parameters: page 2

```
change system-parameters features                              Page   2 of  19
                    FEATURE-RELATED SYSTEM PARAMETERS
LEAVE WORD CALLING PARAMETERS
                              Maximum Number of Messages Per Station: ~0_
                  Maximum Number of External Calls Logged Per Station: 0
                        Message Waiting Indication for External Calls? n
  Stations with System-wide Retrieval Permission (enter extension)
    1: _____________  9: _____________ 17: _____________ 25: _____________
    2: _____________ 10: _____________ 18: _____________ 26: _____________
    3: _____________ 11: _____________ 19: _____________ 27: _____________
    4: _____________ 12: _____________ 20: _____________ 28: _____________
    5: _____________ 13: _____________ 21: _____________ 29: _____________
    6: _____________ 14: _____________ 22: _____________ 30: _____________
    7: _____________ 15: _____________ 23: _____________
    8: _____________ 16: _____________ 24: _____________

                    Prohibit Bridging Onto Calls With Data Privacy? n
                        Enhanced Abbreviated Dial Length (3 or 4): 3
              Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

LEAVE WORD CALLING PARAMETERS

Default Multimedia Outgoing Trunk Parameter Selection

Valid Entry	Usage
2x56 2x64	Sets the default parameter for bandwidth and bearer for all video calls.

Related Links

Multimedia Call Access Code on page 264

Enhanced Abbreviated Dial Length (3 or 4)

Enhanced abbreviated dial lists are for users who need to store a vast quantity of speed-dialed numbers. The administrator might not be able to use all entry slots because of system capacity constraints.

Valid Entry	Usage
3	Administrator can use 1000 enhanced list entries.
4	Administrator can use 10,000 entries available.

Maximum Number of External Calls Logged Per Station

When an external call is unanswered, the server running Communication Manager keeps a record of up to 15 calls, provided information on the caller identification is available, and the telephone message lamp lights. The telephone set displays the names and numbers of unsuccessful callers.

Valid Entry	Usage
0 to 15	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in **Maximum Number of Messages Per Station**.

Maximum Number of Messages Per Station

Valid Entry	Usage
0 to 125	The maximum number of Leave Word Calling (LWC) messages that can be stored by the system for a telephone at a given time. With LWC, internal system users can leave a short pre-programmed message for other internal users. When the message is stored, the Automatic Message Waiting lamp lights on the called telephone.

Message Waiting Indication for External Calls

Enables or disables a station's ability to receive a message waiting indication when external calls are logged.

Prohibit Bridging Onto Calls with Data Privacy

Valid Entry	Usage
y	Prohibits calls from getting bridged by any party, including Service Observing, Intrusion, Verify, and Bridging.
n	Allows calls to be bridged.

Stations With System-wide Retrieval Permission (enter extension)

The server running Communication Manager refers to extensions with system-wide retrieval permission as “super-retrievers”.

Valid Entry	Usage
Assigned extension	Up to ten telephone extension numbers that have permission to retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is disallowed.
attd	All attendants have retrieval permission.

Note:

An extension must be removed from this list before the station can be removed from the system.

Feature-related system parameters: page 3

```
change system-parameters features                              Page   3 of  19
                     FEATURE-RELATED SYSTEM PARAMETERS
TTI/PSA PARAMETERS

  WARNING!  SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

        Terminal Translation Initialization (TTI) Enabled? n




               Customer Telephone Activation(CTA) Enabled? n

                     Hot Desking Enhancement Station Lock? n

EMU PARAMETERS
           EMU Inactivity Interval for Deactivation(hours): __

CALL PROCESSING OVERLOAD MITIGATION
           Restrict Calls: stations-first
```

TTI/PSA PARAMETERS

CPN, ANI for Dissociated Sets

Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets if no system-wide calling party information has been administered for those protocols. A dissociated set is a telephone without an extension. Available only if a default COR has been administered for dissociated sets. Accepts up to 20 digits.

Customer Telephone Activation (CTA) Enabled

Enables or disables the Customer Telephone Activation (CTA) feature that associates a physical telephone with a station extension. CTA applies only to DCP and analog telephones.

Default COR for Dissociated Sets

Available only when Terminal Translation Initialization is enabled for the system.

Valid Entry	Usage
0 to 995 blank	The Class of Restriction (COR) that the system uses for calls made from dissociated telephones. A dissociated set is a telephone without an extension.

Comments on this document? infodev@avaya.com

Enhanced PSA Location/Display Information Enabled

Enables or disables the display of Personal Station Access (PSA) information used to disassociate and associate telephones.

When enabled, displays:

- PSA login and associated station information when a station is PSA associated
- PSA logout and the port when a station is PSA dissociated

Available only when Terminal Translation Initialization (TTI) is enabled for the system.

Hot Desking Enhancement Station Lock

Administers the Hot Desking Enhancement (HDE) feature. “Hot desking” is a feature that can lock and unlock telephones or move a fully customized station profile to another compatible telephone.

Valid Entry	Usage
y	Enables the Hot Desking Enhancement feature.
n	Disables the Hot Desking Enhancement feature.

Terminal Translation Initialization (TTI) Enabled

Technicians use the **Terminal Translation Initialization (TTI) Enabled** field to associate and disassociate telephones. This field is available only if TTI is enabled.

Caution:

Before making changes to the TTI settings, go to the Avaya Support website at http://support.avaya.com/ for current documentation, product notices, knowledge articles.

Valid entry	Usage
y	Starts ACTR, TTI, and PSA transactions for extension and telephone moves between ports.
n	Removes existing TTI port translations and prevents the generation of new TTI port translations.

TTI Security Code

Number that a TTI user uses to access TTI from a telephone or data terminal. Accepts up to seven digits. Available only if Terminal Translation Initialization (TTI) is enabled.

TTI State

The type of port translation used for unadministered digital ports. Available only when Terminal Translation Initialization (TTI) is enabled.

Valid Entry	Usage
data	A stand-alone data module is the TTI port translation for the system. The activation and deactivation sequence is entered at the data terminal.
resume	TTI is available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manually suspended.
suspend	TTI voice or TTI data translations are temporarily unavailable. The system does not remove existing TTI translations.
voice	A voice or voice/data terminal is the TTI port translation for the system. The activation and deactivation sequence is entered from a telephone.

Unnamed Registrations and PSA for IP Telephones

Use this field to allow IP telephones access to the Personal Station Access (PSA) feature. If you set this field to y, IP telephones can register into the PSA dissociated state. The PSA dissociated state is also known as TTI unmerged state , TTI state, and Unnamed Registered for H.323 standards state.

EMU PARAMETERS

EMU Inactivity Interval for Deactivation (hours)

A system-wide administrable interval for EMU de-registration at the visited switch. This timer is applicable to inter and intra-Communication Manager EMU registrations.

Valid Entry	Usage
1 to 24	The interval in hours, after which a visiting user is dropped due to inactivity. Default is 1. An entry of 1 means that after 1 hour of inactivity, the telephone is dropped from the visited home server.
blank	The timer is not used and the visited station remains active until de-registration by another means occurs.

CALL PROCESSING OVERLOAD MITIGATION

Restrict Calls

Indicates the type of calls to block first during overload traffic conditions on the system.

Valid Entry	Usage
stations-first	Deny new traffic generated by internal stations, allowing inbound calls only. This works best in call center environments.
all-trunks-first	Deny all incoming trunk calls and does not block station off-hook.
public-trunks-first	Deny incoming calls on ISDN public trunks and analog CO trunks.

Feature-related system parameters: page 4

```
change system-parameters features                              Page   4 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
         Reserved Slots for Attendant Priority Queue: 5
                           Time before Off-hook Alert: 10
               Emergency Access Redirection Extension:
Number of Emergency Calls Allowed in Attendant Queue: 5
       Drop Parking User From the Call After Timeout? n
   Deluxe Paging and Call Park Timeout to Originator? n
  Controlled Outward Restriction Intercept Treatment: tone
 Controlled Termination Restriction (Do Not Disturb): tone
            Controlled Station to Station Restriction: tone

AUTHORIZATION CODE PARAMETERS
                  Authorization Codes Enabled? n




                 Controlled Toll Restriction Replaces: none
```

Call Pickup on Intercom Calls

Provides or prohibits the system-wide use of Call Pickup or Directed Call Pickup features on intercom calls.

Controlled Outward Restriction Intercept Treatment

The type of intercept treatment the caller receives when the call is outward restricted.

Valid Entry	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. The calling party receives indication that the call is receiving Intercept Treatment. Requires an extension number for the announcement in the associated field. **Note:** If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.

Table continues…

Valid Entry	Usage
extension	Routes to an extension. Requires an extension number for the extension in an associated field. Cannot be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed.

Controlled Station-to-Station Restriction

The type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid Entry	Usage
announcement	Provides an announcement. Requires entry of the announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	Routes the call to an alternate extension. Requires entry of the station or individual attendant. This cannot be a VDN extension.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

The type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid Entry	Usage
announcement	Redirects intercepted calls to an announcement. Requires an associated extension number.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	Redirects intercepted calls to an extension. Requires an associated extension number. Cannot be a VDN extension.
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Enables or disables the Call Park feature so that a user can retrieve a call that is on hold from any other telephone within the system. For example, a user can answer a call at one extension, put the call on hold, and then retrieve the call at another extension. Or the user can answer a call at any telephone after an attendant or another user pages the user. Paged calls that are to be parked require separate activation of the Call Park feature.

Valid Entry	Usage
y	The system redirects a parked call to the user who parked the call.
n	The system redirects a parked call to the attendant.

Drop Parking User From the Call After Timeout

Valid Entry	Usage
y	Drops the parking user from the call automatically after the timeout.
n	Does not drop the parking user from the call automatically after the timeout.

Emergency Access Redirection Extension

The assigned extension number or Vector Directory Number (VDN) where emergency queue overflow redirects.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid Entry	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup is ignored.

Number of Emergency Calls Allowed in Attendant Queue

Valid Entry	Usage
0 to 75	The number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid Entry	Usage
2 to 342	The number of calls that can go in to the emergency queue.

Time Before Off-Hook Alert

Valid Entry	Usage
1 to 3000	The time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

Applies only to remote users or incoming calls over trunks requiring an authorization code. Available only if authorization codes are enabled for the system.

Valid Entry	Usage
y	A call is routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.
n	The caller receives intercept tone.

Related Links

Authorization Codes Enabled on page 284

Authorization Code Cancellation Symbol

The symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. Available only if authorization codes are enabled for the system.

Valid Entry	Usage
(#)	The main and tandem servers or switches are both of the same type.
(1)	An Avaya server or switch is part of the complex or network.

Authorization Code Length

Available only if authorization codes are enabled for the system.

Security alert:

Enhance system security by using the maximum length for your authorization code.

Valid Entry	Usage
4 to 13 digits	The required length of the authorization code. All administered authorization codes must be this length.

Authorization Codes Enabled

Enables or disables authorization codes on a system-wide basis.

Available only if authorization codes are enabled.

Security alert:

To maintain system security, use authorization codes.

Controlled Toll Restriction Intercept Treatment

Applies an intercept treatment to a toll call during the call processing. Available only if the Controlled Toll Restriction feature is activated.

Valid Entry	Usage
announcement	Intercepted calls are redirected to an announcement. Requires an assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	Intercepted calls are redirected to an extension. Requires an extension assigned to a station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Toll Restriction Replaces

Activates the Controlled Toll Restriction feature for the customer to use additional types of calling restrictions on guest room telephones.

Valid Entry	Usage
outward	Guests cannot place calls to the public network.
station-station	Guests cannot place or receive calls between guest rooms or administrative staff voice terminals.
termination	Guests cannot receive any calls.
total	Guests cannot place or receive any calls.
toll	Guests cannot place toll calls, but can place local free calls.

Display Authorization Code

Enables or disables displaying authorization code digits on the set during dialing. Applies only to DCP, not to BRI or hybrid sets.

Security alert:

To enhance system security, do *not* display authorization code digits during dialing.

Feature-related system parameters: page 5

```
change system-parameters features                              Page   5 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint: Preserve       Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                     Switch Name:
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                            COR to Use for DPT: station
                 EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
                Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
      Delay Sending RELease (seconds): 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n
              Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? n    UCID Network Node ID:
```

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

Endpoint

Valid Entry	Usage
SYS_PRNT	If the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT for IP Services
blank	Not administered

Lines Per Page

Valid Entry	Usage
24 to 132	The number of lines per page required for the report.

SYSTEM-WIDE PARAMETERS

COR to Use for DPT

The Class of Restriction used for the Dial Plan Transparency (DPT) feature.

Valid Entry	Usage
station	The Facility Restriction Level (FRL) of the calling station determines whether that station is permitted to make a trunk call and if so, which trunks it is eligible to access. This is the default.
unrestricted	The first available trunk preference pointed to by ARS routing is used.

Emergency Extension Forwarding (min)

If an emergency call disconnects, public safety personnel always attempt to call back. If the ELIN that was sent differs from the caller's extension number, the return call rings a set other than the one that dialed 911. To overcome this limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same switch as the extension that dialed 911. Customers who have several switches in a campus should assign emergency location extensions accordingly.

Sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off.

Valid Entry	Usage
0 to 999	The time in minutes that an incoming trunk call forwards to the extension that made the initial 911 call. The default value for both new installs and upgrades is 10.

Note:

If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes remain on the timer.

Enable Dial Plan Transparency in Survivable Mode

Enables or disables Dial Plan Transparency (DPT) without changing or removing other feature administration associated with DPT. DPT is enabled if a gateway registers with a Survivable Remote Server (Local survivable processor), or a port network registers with a Survivable Core Server (Enterprise Survivable Server).

Enable Inter-Gateway Alternate Routing

Enables or disables the Inter-Gateway Alternate Routing (IGAR) feature that provides a means of alternately using the public network when the IP-WAN is incapable of carrying the bearer connection.

IGAR Over IP Trunks

The system displays the field when you set the **Enable Inter-Gateway Alternate Routing** field to y.

Valid Entry	Usage
allow	The IGAR feature uses H.323 or SIP trunks when selecting a trunk from a Route Pattern.
skip	The IGAR feature skips H.323 and SIP trunks when selecting a trunk from a Route Pattern. By default, the value is skip.

Switch Name

A name used for identifying the switch. Accepts up to 20 alphanumeric characters.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Enables or disables an audible tone to the controlling station when a Malicious Call Trace (MCT) recorder is actively recording a malicious call.

Delay Sending Release (seconds)

Available only if Malicious Call Trace is enabled for the system.

Valid Entry	Usage
0 to 30	The time in seconds the system waits before sending an ISDN release message in response to receiving an ISDN disconnect message. Available in increments of 10.

MCT Voice Recorder Trunk Group

Valid Entry	Usage
1 to 2000	Assigns the trunk group number for Malicious Call Trace (MCT) voice recorders.

SEND ALL CALL OPTIONS

Auto Inspect on Send All Calls

Valid Entry	Usage
y	The system displays Calling Party information for calls that are silently alerting their station because of the Send-All-Calls feature.
n	Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature is not guaranteed.

Send All Calls Applies to

Valid Entry	Usage
station	Any call to a station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to have Ring-Ping notification if redirect notification is enabled.
extension	Only the calls sent to that extension are placed to coverage.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Enables or disables the generation of a UCID for each call when necessary.

UCID Network Node ID

Valid Entry	Usage
1 to 327 blank	A number unique to this server or switch in a network of switches. This number is an important part of the UCID tag and must be unique to the server or switch.

Feature-related system parameters: page 6

```
change system-parameters features                              Page   6 of  19
                     FEATURE-RELATED SYSTEM PARAMETERS
         Public Network Trunks on Conference Call: 5                Auto Start? n
   Conference Parties with Public Network Trunks: 6                  Auto Hold? n
Conference Parties without Public Network Trunks: 6               Attendant Tone? y
        Night Service Disconnect Timer (seconds): 180              Bridging Tone? n
                 Short Interdigit Timer (seconds): 3            Conference Tone? n
              Unanswered DID Call Timer (seconds):              Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30     Mode Code Interface? y
                 Long Hold Recall Timer (seconds): 0
                      Reset Shift Timer (seconds): 0
    Station Call Transfer Recall Timer (seconds): 0           Recall from VDN? n
          Trunk Alerting Tone Interval (seconds): 15
                              DID Busy Treatment: tone
              Allow AAR/ARS Access from DID/DIOD? n
                Allow ANI Restriction on AAR/ARS? n
Use Trunk COR for Outgoing Trunk Disconnect/Alert? n
                 7405ND Numeric Terminal Display? n                    7434ND? n



   DTMF Tone Feedback Signal to VRU - Connection:          Disconnection:
```

7405ND Numeric Terminal Display

Enables or disables allowing a 7405ND type of station. This is not an actual telephone type, but it can be used to define ports for certain types of voice messaging systems. This numeric display setting sends only numbers, and not names, to the messaging system.

Related Links

Type on page 742

7434ND

Enables or disables allowing a 7434ND type of station. This is not an actual telephone type, but it can be used to define ports for certain types of messaging systems. Used if the voice messaging system operates in Bridged Mode.

Related Links

Type on page 742

Allow AAR/ARS Access from DID/DIOD

Enables or disables allowing calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only). Enables or disables allowing a call placed over a Russian shuttle trunk or a Russian rotary trunk using AAR or ARS to have the ANI requirement administered as restricted. In this case, when ANI is requested, if the request fails, the call immediately drops.

Related Links

ANI Reqd on page 17
ANI Reqd on page 22
Request Incoming ANI (non-AAR/ARS) on page 577

Attendant Tone

Enables or disables providing call progress tones to the attendants.

Auto Hold

Enables or disables the Automatic Hold feature on a system-wide basis.

Auto Start

If enabled, the Start buttons on all attendant consoles are disabled and the Automatic Start feature is enabled.

Bridging Tone

Enables or disables providing a bridging tone when calls are bridged on primary extensions.

Conference Parties without Public Network Trunks

Valid Entry	Usage
3 to 6	The maximum number of parties allowed in a conference call involving no public network trunks.

Conference Parties with Public Network Trunks

Available only if public network trunks are allowed on a conference call.

Valid Entry	Usage
3 to 6	The maximum number of parties allowed in a conference call involving a public network subscriber.

Related Links

Public Network Trunks on Conference Call on page 293

Conference Tone

Enables or disables providing a conference tone as long as three or more calls are in a conference call.

Note:

Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones are equivalent to no tone (silence) unless the tone is independently administered or customized.

Related Links

Tone Generation on page 856

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid Entry	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

Available only if DTMF Feedback Signals for the Voice Response Unit (VRU) are enabled for the system.

Valid Entry	Usage
0 to 9, *, #, A, B, C, D	The code used to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	No tone is sent to the VRU.

Related Links

DTMF Feedback Signals For VRU on page 808

Intrusion Tone

Enables or disables applying an intrusion tone (executive override) when an attendant intrudes on the call.

Line Intercept Tone Timer (seconds)

Valid Entry	Usage
0 to 60	How long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

Valid Entry	Usage
0 to 999	The number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Activates or deactivates the use of the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to other voice mail systems.

Note:

After making changes, log off and log back on to access the **Mode Code Related System Parameters**.

Night Service Disconnect Timer (seconds)

Valid Entry	Usage
10 to 1024 blank	The number of seconds a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public Network Trunks on Conference Call

Valid Entry	Usage
0 to 5	The number of public network trunks allowed on a conference call.

Recall from VDN

Indicates whether or not a call that is transferred to a VDN and then routed to a station is recalled to the originating station after the Station Call Transfer Recall Timer expires. If enabled, calls are recalled from a VDN when the Station Call Transfer Recall Timer expires.

Available only if Basic Vectoring and Vectoring (Prompting) are enabled for the system.

Related Links

Vectoring (Basic) on page 810
Vectoring (Prompting) on page 811
Station Call Transfer Recall Timer (seconds) on page 294

Reset Shift Timer (seconds)

Valid Entry	Usage
0 to 255	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. With reset shift dial tone, the user can dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. An entry of 0 disables this feature. Used only for station-to-station calls or private network calls using ISDN trunks.

Short Interdigit Timer (seconds)

Valid Entry	Usage
3 to 9	The time limit that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

If enabled, Special Dial Tone can be used.

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Requires a TN2182 circuit pack, or any newer tone generation resource.

Related Links

Special Dial Tone on page 331

Station Call Transfer Recall Timer (seconds)

A user-transferred call (station-to-station, a trunk call, or a DCS call) can re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid Entry	Usage
0 to 999	The time in seconds before a call redirects back to the station user who initiated the transfer operation. The entry 0 disables this feature.

Trunk Alerting Tone Interval (seconds)

Valid Entry	Usage
1 to 60	Specifies the interval at which the alerting tone is repeated on the call. By default, the value is 15 seconds.

Related Links

Outgoing Trunk Alerting Timer (minutes) on page 126

Unanswered DID Call Timer (seconds)

Valid Entry	Usage
10 to 1024	Limits how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment. This timer interacts with the nonadministrable 50-second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Therefore, if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored.
blank	Disables the timer.

Related Links

Wait Answer Supervision Timer on page 312

Use Trunk COR for Outgoing Trunk Disconnect/Alert

Indicates whether the outgoing trunk disconnect timer or the outgoing trunk alerting timer is set based on the COR of the originating station or the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating station. By default, this field is disabled.

Important:

You must disable outgoing trunk disconnect and outgoing trunk alerting timers at the same time. If you try to administer both timers, the system displays the following message:

```
Cannot enable both Outgoing Trunk Disc and Outgoing Trunk Alert timers
```

Related Links

Outgoing Trunk Disconnect Timer (minutes) on page 126
Outgoing Trunk Alerting Timer (minutes) on page 126

DISTINCTIVE AUDIBLE ALERTING

Available only if **Tenant Partitioning** is disabled for the system.

Attendant Originated Calls

Indicates which type of ringing applies to attendant-originated calls.

Valid Entry	Usage
internal	Internal ringing applies to attendant-originated calls.

Table continues…

Valid Entry	Usage
external	External ringing applies to attendant-originated calls. Default is external.
priority	Priority ringing applies to attendant-originated calls.

Distinctive Audible Alerting (Internal, External, Priority)

The number of rings for Internal, External, and Priority calls. For virtual stations, this applies to the mapped-to physical telephone.

Valid Entry	Usage
1	One burst of ringing signal per period. Default for internal calls.
2	Two bursts of ringing signal per period. Default for external and attendant calls.
3	Three bursts of ringing signal per period. Default for priority calls.

Related Links

Distinctive Audible Alerting (Internal, External, Priority) on page 834

Feature-related system parameters: page 7

```
change system-parameters features                              Page   7 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER
               Abort Transfer? n                  No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n        Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                    Unhold? n
   No Hold Conference Timeout: 60

                  External Ringing for Calls with Trunks? remote-only
ANALOG BUSY AUTO CALLBACK
               Without Flash? n



AUDIX ONE-STEP RECORDING
                            Recording Delay Timer (msec): 500
Apply Ready Indication Tone To Which Parties In The Call? all
  Interval For Applying Periodic Alerting Tone (seconds): 15
                                 Audix Recording Display? n
POSTED MESSAGE
        Require Security Code? n
```

CONFERENCE/TRANSFER

Abort Conference

DCP, hybrid, IP, wireless, or ISDN-BRI telephone users can abort the conference operation when they are disconnected. If enabled, this field changes a call placed on soft-hold in the conference-pending status to hard-held status if the user disconnects.

Abort Transfer

If enabled, stops the transfer operation whenever users press a non-idle call appearance button in the middle of the transfer operation, or when they disconnect. If the system is configured to transfer calls upon disconnecting the call, users can press the **Transfer** button, dial the complete transfer-to number, and disconnect to transfer the call. Users must select another non-idle call appearance to abort the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless telephones.

External Ringing for Calls with Trunks

Specifies ringing behavior on external trunk calls that are transferred or conferenced by stations or attendants, or extended by the attendant to an on-switch extension.

Valid Entry	Usage
all-calls	All external trunk calls that are transferred or conferenced (either locally or remotely) receive external ringing.
local-only	External trunk calls that are transferred or conferenced locally receive external ringing.
none	External ringing does not apply to external trunk calls that are transferred or conferenced.
remote-only	External trunk calls that are transferred or conferenced remotely receive external ringing. This is the default value.

Maximum Ports per Expanded Meet-me Conf

Available only if ports are administered for the Expanded Meet-me Conference feature.

Valid Entry	Usage
3 to 300	The maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit (that is, not administrable on a per Expanded-Meet-me VDN basis).

No Dial Tone Conferencing

If enabled, eliminates dial tone while setting up a conference when another line is on hold or is alerting.

No Hold Conference Timeout

Valid Entry	Usage
20 to 120	The number of seconds No Hold Conference call setup times out. The system **Answer Supervision** timer should be set to a value less than this.

Select Line Appearance Conferencing

User can use the line appearance rather than the **Confererence** button to include a call in a conference. If a user is on a call, and another line is on hold or an incoming call alerts on another line, the user can press the **Confererence** button to bridge the calls together. Using the select line appearance capability, the user can press a **line appearance** button to complete a conference instead of pressing the **Confererence** button a second time.

Enabling this field activates Select Line Appearance Conferencing.

Transfer Upon Hang-Up

With this feature, users can transfer a call by pressing the **Transfer** button, dialing the required extension, and then disconnecting the active call.

Users using the **Call Park** FAC can park a call without pressing the **Transfer** button a second time.

Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to 96x1 SIP stations. However, for SIP stations, this field works only when the Team button feature is enabled.

Unhold

With unhold, the user can press the hold button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI telephones or attendant consoles.

Enabling this field activates this unhold capability.

ANALOG BUSY AUTO CALLBACK

If the Analog Busy Auto Callback Without Flash (ACB) feature is enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path or forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter 1 for ACB or 2 to cover to a voice mail hunt group extension.

Announcement

The extension of the announcement that plays for the Analog Busy Auto Callback Without Flash feature. This field cannot be left blank. Available only if Auto Callback Without Flash is enabled.

Voice Mail Hunt Group Ext

A voice mail hunt group extension where the call is forwarded if the user enters `2` at the ACB announcement prompt. Available only if Auto Callback Without Flash is enabled.

Without Flash

If enabled, provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

AUDIX ONE-STEP RECORDING

On stations administered with this feature button, users can activate and deactivate the recording of active calls to their voice messaging system with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

Use this field to administer who hears the ready indication tone.

Valid entry	Usage
all	All parties in the call hear the tone. The default value is all.
initiator	Only the initiator hears the tone. The initiator is the user who activates the Audix one-step recording feature.
none	None of the parties in the call hear the tone.

Note:

When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to `initiator`, all parties in the call hear the ready indication tone. When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to `all` and the **Interval For Applying Periodic Alerting Tone (seconds)** field is set to `0`, the parties in the call do not hear the ready indication tone and the alerting tone.

Interval For Applying Periodic Alerting Tone (seconds)

Available only if all parties on a call hear the voice messaging recording ready tone.

Valid Entry	Usage
0 to 60	The number of seconds between alerting tones. The default value is a 15 second interval. Zero disables the tone.

Recording Delay Timer (msecs)

Valid Entry	Usage
0 to 4000 in increments of 100	The delay interval before starting a voice messaging recording.

Feature-related system parameters: page 8

```
change system-parameters features                              Page   8 of  19
                     FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS
                                                  PARAMETERS FOR CREATING
 Send Non-ISDN Trunk Group Name as Connected Name? n     QSIG SELECTION NUMBERS
 Display Connected Name/Number for ISDN DCS Calls? n       Network Level: _
       Send ISDN Trunk Group Name on Tandem Calls? n        Level 2 Code: _____
                 Send Custom Messages Through QSIG? n       Level 1 Code: _____
                           QSIG/ETSI TSC Extension: _____________
 MWI - Number of Digits Per Voice Mail Subscriber: 7

                              National CPN Prefix: _____
                          International CPN Prefix: _____
                                Pass Prefixed CPN: ASAI? n   VDN/Vector? n
     Delay for USNI Calling Name for Analog Caller ID Phones (seconds): 0
   Unknown Numbers Considered Internal for AUDIX? n
             USNI Calling Name for Outgoing Calls? n
               Path Replacement with Measurements? y
                  QSIG Path Replacement Extension: _____________
  Send QSIG Path Replacement Conf. Event to ASAI? y
           Path Replace While in Queue/Vectoring? n
```

ISDN PARAMETERS

Delay for USNI Calling Name for Analog Caller ID Phones (seconds)

For analog caller ID telephones, Communication Manager waits for the FACILITY message (includes the caller information) before delivering an incoming call to the receiver.

Valid Entry	Usage
0 to 6	Communication Manager waits for the administered duration for the FACILITY message before delivering the incoming call to the receiver. The default value of the field is 0 which indicates that the feature is disabled. Use the timer value of 3 seconds.

Note:

On page 3 of the ISDN Trunk Group screen, you must set **US NI Delayed Calling Name Update** field to y for this feature to work.

Display Connected Name/Number for ISDN DCS Calls

If enabled, displays the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

An extension used for proper termination of Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension to send the indication to the appropriate server running Communication Manager. Available only if ISDN Feature Plus signaling is enabled for the system.

International CPN Prefix

You can use this number to apply prefixes to international calling numbers for display at receiving telephones. Accepts up to five digits and includes the characters * and #. This number is useful for those telephones that use or implement call back features based on incoming call numbers.

When an ISDN-PRI call arrives, the incoming call setup is analyzed for:

- Whether the Type of Address (TOA) is national or international
- Whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony

This administered prefix is applied to international calls.

Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Maximum Length

Available only if unknown numbers are administered as internal for the voice messaging system.

Valid Entry	Usage
1 to 20	The maximum length of an unknown private number. Any unknown number longer than the administered value is external. This field cannot be blank.

MWI - Number of Digits Per Voice Mail Subscriber

The number of digits per voice messaging system subscriber. Available only if Basic Supplementary Services or ISDN Feature Plus signaling is enabled for the system.

Valid Entry	Usage
3 to 7	The digit string length of subscribers translated in the Message Center entity. The value in this field must match the value administered for the voice messaging system.

National CPN Prefix

Applies prefixes to national calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony.

This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid Entry	Usage
0 to 9 * or # blank	A five-digit number that you can apply as prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

If enabled, passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Path Replacement While in Queue/Vectoring

Use this field to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS, some calls can go unrecorded if this capability is enabled. For more information, go to Avaya Support website at http://support.avaya.com/.

Path Replacement with Measurements

Use this field to allow QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG/ETSI TSC Extension

The phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs) that are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing). ETSI protocol TSCs as well as QSIG TSCs are supported.

QSIG Path Replacement Extension

The extension for the system used as part of the complete number sent in the Path Replacement Propose message.

Send Custom Messages Through QSIG?

If enabled, provides appropriate display information over QSIG links. Display information can include the Posted Messages feature.

Send ISDN Trunk Group Name on Tandem Calls

If enabled, provides consistent display information regardless of trunk type. Also provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

If enabled, sends a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Controls the treatment of an ISDN number whose numbering plan identification is “unknown” in a QSIG centralized voice messaging system arrangement. Available only if ISDN trunks are enabled for the system and if the hunt group is administered to send the calling party number to the voice messaging system.

Valid Entry	Usage
y	The unknown number is “internal” and the voice messaging system tries to find a calling party name match for the digit string. If a name match is found, the voice messaging system provides the calling party’s name. If no name is found, the voice messaging system provides the calling party’s telephone number.
n	The unknown number is “external” and the voice messaging system provides the calling party’s telephone number.

USNI Calling Name for Outgoing Calls?

Valid Entry	Usage
y	Sends a name on outgoing calls over NI PRI trunks.
n	Prevents sending calling name information with outgoing calls over NI PRI trunks. Overrides the trunk group **Send Name** if enabled.

Caution:

The service provider’s local telephone company central office (CO) must be capable of accepting calling name information from Communication Manager in this way. For example, if the CO has a 5ESS, it must be a generic 5EXX or later. Failure to validate the CO capability might cause the CO to drop outgoing calls from the Avaya S8*XXX* Server. In this case, this field should be disabled.

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS

Level 1 Code

The first level regional code of the Avaya S8*XXX* Server in the network.

Administer this field carefully. Communication Manager does not verify if you have entered a code that supports your entry in the **Network Level** field. Accepts up to five digits. Because blank regional codes are valid, an entry is not required if the **Network Level** field is `1` or `2`.

In QSIG standards, this Level 1 code is called the Level 0 Regional Code.

Available only if the **Network Level** is set to 1 or 2.

Level 2 Code

The second level regional code of the Avaya S8*XXX* Server in the network.

Administer this field carefully. Communication Manager does not check to ensure you have entered a code that supports your entry in the **Network Level** field. Accepts up to five digits. Because blank regional codes are valid, an entry is not required if the **Network Level** field is 2.

Available only if the **Network Level** is set to 2.

In QSIG standards, this Level 2 code is called the Level 1 Regional Code.

Network Level

The value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/ Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid Entry	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the **Send Calling Number and/or Send Connected Number** field is y or r with private specified for the **Numbering Format** field on the ISDN Trunk Group screen, the Calling Number and/or Connected Number IEs is not sent. If the field is left blank but private has been specified in the **Numbering Format** field on the ISDN Trunk Group screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as PresentationRestricted with NULL for the associated digits.

Feature-related system parameters: page 9

```
change system-parameters features                              Page   9 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: _______________
  CPN/ANI/ICLID Replacement for Unavailable Calls: _______________

DISPLAY TEXT
                                       Identity When Bridging: principal
                                        User Guidance Display? n
 Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code: ___
          International Access Code: _____

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
     Caller ID on Call Waiting Delay Timer (msec): 200
```

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Use this field to specify a text string to replace the restricted numbers on the telephone display. You can enter up to 15 characters.

Note:

This field does not apply to SIP telephones.

CPN/ANI/ICLID Replacement for Unavailable Calls

Use this field to specify a text string to replace the unavailable numbers on the telephone display. You can enter up to 15 characters.

Note:

This field does not apply to SIP telephones.

DISPLAY TEXT

Extension only label for Team button on 96xx H.323 terminals

Available only if the **Team Btn Display Name** for the Class of Restriction is enabled.

Valid entries	Usage
y	For 96xx H.323 telephones, displays the station extension without the team button label. 96xx H323 firmware version 2.0 or greater is recommended as it provides a special icon for team buttons. This field does not impact label customizations.
n	For 96xx H.323 telephones, displays the station extension with the team button label.

Identity When Bridging

Determines whether the telephone display shows the identity of the extension that the bridged appearance represents, or the identity of the physical telephone that the bridged appearance button is on. This applies when the bridged appearance is the caller, and when it is the answering party.

Note:

When you choose the station option, you must update the Public/Unknown Numbering Format with the Extension Codes of the stations that display the caller's or answering party's assigned identification.

Valid Entry	Usage
principal	The extension that the bridged appearance button points to. This is the default.
station	The physical telephone that the bridged appearance button is on.

User Guidance Display

If enabled, the telephone display shows user guidance messages. This field is disabled by default.

INTERNATIONAL CALL ROUTING PARAMETERS

Local Country Code

A valid PSTN E.164 country code. For example, an SBS node in the United States uses a code of 1. For a list of country codes, see the International Telecommunications Union *List of ITU-T Recommendation E.164 Assigned Country Codes*. Accepts up to three digits.

Related Links

SBS on page 503
International Access Code on page 307
Group Type on page 482
Supplementary Service Protocol on page 495

Carrier Medium on page 479

International Access Code

The access code required by the PSTN to route calls out of the country. This code is included with the telephone number received from the SBS terminating node if the Local Country Codes of the originating and terminating nodes are different. For example, an SBS node in the United States uses an access code of `011`. Accepts up to five digits.

Note:

Once administered, this field cannot be cleared until all trunk groups administered for SBS signaling have been removed. If the international call routing parameters are not administered for the system and SBS is enabled for the trunk, a warning is displayed: `Must set INTERNATIONAL CALL ROUTING parameters on system-parameters features.`

Related Links

SBS on page 503
Local Country Code on page 306
Group Type on page 482
Supplementary Service Protocol on page 495
Carrier Medium on page 479

SCCAN PARAMETERS

Enable Enbloc Dialing without ARS FAC

If the **Enable Enbloc Dialing without ARS FAC** field is enabled, Communication Manager inserts the ARS access code, when a dialed digit string has:

- **Minimum Digit Length** or more digits, and is sent from the telephone to Communication Manager in a single message
- a matching ARS entry

If the dialed digit string does not meet above points, the digit string is used as it is without prepending the ARS access code. The call is passed to ARS for digit analysis. By default, the **Enable Enbloc Dialing without ARS FAC** field is disabled.

Minimum Digit Length

Available only if Enbloc Dialing without an ARS FAC is enabled.

Valid Entry	Usage
1 to 20	The number of digits before Enbloc Calling Treatment is activated. Default is extension length plus 1.

CALLER ID ON CALL WAITING PARAMETERS

Caller ID on Call Waiting Delay Timer (msec)

Valid Entry	Usage
5 to 1275	The required delay in 5-millisecond intervals. Default is 200.

Feature-related system parameters: page 10

```
change system-parameters features                              Page  10 of  19
                     FEATURE-RELATED SYSTEM PARAMETERS

               Pull Transfer: n              Update Transferred Ring Pattern? n
        Outpulse Without Tone? y                Wait Answer Supervision Timer? n
        Misoperation Alerting? n                 Repetitive Call Waiting Tone? n
   Allow Conference via Flash? y
Vector Disconnect Timer (min): ___     Network Feedback During Tone Detection? y
  Hear Zip Tone Following VOA? y     System Updates Time On Station Displays? n

                  Station Tone Forward Disconnect: silence
                           Level Of Tone Detection: precise
         Charge Display Update Frequency (seconds): 30
                        Date Format on Terminals: mm/dd/yy
                     Onhook Dialing on Terminals? n
             Edit Dialing on 96xx H.323 Terminals? n
                Allow Crisis Alert Across Tenants? n


ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? n
```

Allow Conference via Flash

If enabled, an analog station can use flash to conference calls.

Allow Crisis Alert Across Tenants

If enabled, allows crisis alert across different tenant partitions. Disabled by default.

Charge Display Update Frequency (seconds)

Available only if Advice of Charge or Periodic Pulse Metering with display functions is used.

Valid Entry	Usage
10 to 60 blank	The amount of time in seconds between charge-display updates. Frequent display updates might have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display does not automatically show charge information.

Date Format on Terminals

The system displays the **Date Format on Terminals** field if the **Multinational Locations** field on the System Parameters Customer Options screen is set to n. This field affects the 607/2400/4600/6400 terminals. The date is in `mm/dd/yy`, `dd/mm/yy`, and `yy/mm/dd` formats.

Edit Dialing on 96xx H.323 Terminals

If enabled, an end-user can dial a number when the telephone is on-hook. Disabled by default.

Hear Zip Tone Following VOA?

Enables or disables zip tone alerts to a telephone user when the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

Note:

This field does not effect auto-answer zip tone heard prior to the VDN of Origin Announcement (VOA).

Intercept Treatment on Failed Trunk Transfers

Valid Entry	Usage
y	Provides intercept treatment to calls failing trunk transfers.
n	Drops calls failing trunk transfers.

Level of Tone Detection

Used only when users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid Entry	Usage
broadband	This is the least exact of the levels of tone detection. If Avaya Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Avaya Communication Manager interprets any tone which has a continuous “on” period of longer than 1 second as dial tone. Otherwise, the server accepts whatever the tone detector circuit pack reports.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Valid Entry	Usage
y	Enables misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Uses standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

If enabled, provides audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

If enabled, users can dial without lifting the handset. Users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Outpulse Without Tone

Valid Entry	Usage
y	The server outpulses digits even when a dial tone has not been received.
n	The calling party receives intercept tone if no dial tone is detected.

Pull Transfer

Enables or disables the Pull Transfer feature on a system-wide basis. With this, the transferring or transferred-to party can press the **Transfer** button to complete the transfer operation.

Repetitive Call Waiting Interval (sec)

Available only if the Repetitive Call Waiting Tone is enabled.

Valid Entry	Usage
1 to 99	The number of seconds between call waiting tones.

Repetitive Call Waiting Tone

If enabled, a repetitive call waiting tone is provided to the called party for all types of call waiting access.

Send DTMF Over Telecommuter Link

In the telecommuter mode, when a user is on a call and presses any digits from a deskphone or a softphone, the user hears loud DTMF tones over the telecommuter link. Use the **Send DTMF Over Telecommuter Link** field to send or prevent DTMF tones over the telecommuter link.

Valid entry	Usage
y	Communication Manager sends DTMF tones over the telecommuter link. The default value is y.
n	Communication Manager does not send DTMF tones over the telecommuter link.

Station Tone Forward Disconnect

Applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid Entry	Usage
busy intercept silence	The treatment given to a station that is the last party remaining off-hook on a call, until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

If enabled, automatically updates the system time on two-line display telephones when you run background maintenance, for example, when you plug in the telephone set. The system sends the time update periodically to the telephones. This parameter does not apply to telephones such as the BRI telephones, where the user sets the time.

Update Transferred Ring Pattern

If enabled, changes the ringing pattern from internal to external when an internal station transfers an external call. Use this feature if most calls go through an attendant, so that users are able to distinguish an external call.

Vector Disconnect Timer (min)

Valid Entry	Usage
1 to 240	Determines when the trunk gets disconnected if incoming or outgoing disconnect supervision is disabled for the trunk group.
blank	Avaya Communication Manager does not initiate a disconnect.

Wait Answer Supervision Timer

Valid Entry	Usage
y	Calls to stations unanswered after 50 seconds are dropped.
n	Unanswered calls drop only when the calling party goes on-hook.

Related Links

Unanswered DID Call Timer (seconds) on page 295

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Apply Intercept Locally

If enabled, DID/CO intercept treatment is applied locally instead of on the originating server or switch. Available only if **Italian Protocol** is enabled.

Enforce PNT-to-PNT Restrictions

If enabled, restrictions and denial of PNT-to-PNT connections are enforced when the EDCS message is unavailable. Available only if **Italian Protocol** is enabled.

Italian Protocol Enabled

Enables or disables the Italian DCS feature on a system-wide basis.

Feature-related system parameters: page 11

```
change system-parameters features                              Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? n
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                  Delay:
    Message Waiting Lamp Indicates Status For: station

  VECTORING
                    Converse First Data Delay: 0      Second Data Delay: 2
                Converse Signaling Tone (msec): 100        Pause (msec): 70
                     Prompting Timeout (secs): 10
                  Interflow-qpos EWT Threshold: 2
    Reverse Star/Pound Digit For Collect Step? n
          Available Agent Adjustments for BSR? n
                              BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
               Service Observing: Warning Tone? y     or Conference Tone? n
 Service Observing/SSC Allowed with Exclusion? n
              Allow Two Observers in Same Call? n
```

CALL CENTER SYSTEM PARAMETERS EAS

Delay

Available only if Expert Agent Selection (EAS) or ASAI Link Core Capabilities are enabled for the system.

Valid Entry	Usage
0 to 99 blank	The number of seconds the caller hears ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

The extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

Enables or disables Expert Agent Selection (EAS). Enable only if ACD or vectoring hunt groups exist or if existing ACD or vectoring hunt groups are skill-based. Available only if EAS is enabled for the system.

Message Waiting Lamp Indicates Status For

Available only if Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
station	The message waiting lamp on a telephone indicates the message is for the telephone extension.
loginID	The message waiting lamp on a telephone indicates the message is for the agent login ID.

Minimum Agent-LoginID Password Length

The minimum number of digits that must be administered as an EAS Agent's LoginID password. Accepts up to nine digits. Available only if Expert Agent Selection (EAS) is enabled for the system.

VECTORING

Available Agent Adjustments for BSR

If activated, Best Service Routing (BSR) adjustments are made available to available agents. Available only if Vectoring (Best Service Routing) is enabled for the system.

BSR Tie Strategy

Available only if Vectoring (Best Service Routing) is enabled for the system.

Valid Entry	Usage
1st-found	BSR uses the first selection for routing. This is the default.
alternate	Alternating the BSR selection algorithm is possible when a tie in Expected Wait Time (EWT) or available agent criteria occurs. Every other time a tie occurs for calls from the same active Vector Directory Number (VDN), the selection from the step with the tie is used instead of the first selected split/skill or location to send the call. This helps balance the routing over the considered local splits/skills and remote locations when cost of routing remotely is of least concern.

Converse First Data Delay

Available only if Vectoring (Basic) is enabled for the system.

Valid Entry	Usage
0 to 9	The number of seconds data is prevented from being outpulsed as a result of a converse vector step from the system to Avaya IR before IR is ready. The delay commences when the IR port answers the call.

Converse Second Data Delay

Available only if Vectoring (Basic) is enabled for the system.

Valid Entry	Usage
0 to 9	Number of seconds used when two groups of digits are being outpulsed, as a result of a converse vector step, from the system to Avaya IR. Prevents the second set from being outpulsed before IR is ready. The delay commences when the first group of digits has been outpulsed.

Converse Signaling Pause

The length in milliseconds of the delay between digits being passed. The optimum timer settings for Avaya IR are 60 msec tone and 60 msec pause.

Available only if Vectoring (Basic) and DTMF feedback signals are enabled for the system.

Valid Entry	Usage
40 to 2550 (in increments of 10)	Values are rounded up or down depending upon the type of circuit pack used to outpulse the digits. • **TN742B or later suffix analog board** — Rounds up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone • **TN464F, TN767E or later suffix DS1 boards** — Rounds up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signaling to IR, and has then been used to send digits to a different destination, IR timers might stay in effect. To reset the timers to the system default, pull and reseat the circuit pack.

Converse Signaling Tone

The length in milliseconds of the digit tone for digits being passed to Avaya IR. The optimum timer settings for IR are 60 msec tone and 60 msec pause.

Available only if Vectoring (Basic) and DTMF feedback signals are enabled for the system.

Valid Entry	Usage
40 to 2550 (in increments of 10)	Values are rounded up or down depending upon the type of circuit pack used to outpulse the digits. • **TN742B or later suffix analog board** — Rounds up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone

Valid Entry	Usage
	• **TN464F, TN767E or later suffix DS1 boards** — Rounds up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signaling to IR, and has then been used to send digits to a different destination, IR timers might stay in effect. To reset the timers to the system default, pull and reseat the circuit pack.

Interflow-qpos EWT Threshold

Available only if the Lookahead Interflow (LAI) is enabled for the system.

Valid Entry	Usage
0 to 9 blank	Number of seconds for this threshold. Any calls predicted to be answered before this threshold are not interflowed, therefore saving CPU resources.

Prompting Timeout (secs)

Available only if Vectoring (Prompting) is enabled for the system.

Valid Entry	Usage
4 to 10	The number of seconds before the **`Collect Digits`** command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The “*” is interpreted as a caller end-of-dialing indicator and the “#” is an indicator to clear all digits previously entered by the caller for the current collect vector step.

Valid Entry	Usage
y	Reverses the star and pound digits for the collect vector step. This does not affect any other vector step or other non-ACD feature, such as ARS.
n	The “*” and “#” digit-processing is unchanged.

Store VDN Name in Station’s Local Call Log

Enables or disables the sending of a message from Communication Manager telling the telephone to store the VDN name or the calling party’s name in the station call log for any of the following telephones:

- 2420
- 4610
- 4620
- 4625

SERVICE OBSERVING

Allow Two Observers in Same Call

Valid Entry	Usage
y	Two service observers can check the same EAS Agent LoginID or station extension, and up to two service observers can be on the same two-party call or in a conferenced call having more than two parties.
n	Only one service observer can check the EAS Agent LoginID or station extension.

Service Observing: Warning Tone

Enables or disables a warning tone that is heard by telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. Available only if Service Observing (Basic) is enabled for the system.

Warning:

The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations — or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Service Observing/SSC Allowed with Exclusion

With **Service Observing/SSC Allowed with Exclusion**, you can use the Service Observing feature or the Single Step Conference (SSC) feature on a station for which **Exclusion** is active. For more information about SSC, see *Avaya Aura® Call Center Elite Feature Reference* .

Valid entry	Usage
y	Communication Manager allows Service Observing or SSC on the station.
n	Communication Manager does not allow Service Observing or SSC on the station. If an agent whose station is being observed activates the **Exclusion** button, Communication Manager drops all the bridged parties, including the observer. The default value is n.

or Conference Tone

Enables or disables a conference tone heard by telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. Available only if Service Observing (Basic) is enabled for the system. Not available if a warning tone is administered for Service Observing.

Warning:

The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations — or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Feature-related system parameters: page 12

```
change system-parameters features                              Page  12 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

 AGENT AND CALL SELECTION
                          MIA Across Splits or Skills? n
                           ACW Agents Considered Idle? y
                           Call Selection Measurement: current-wait-time
   Service Level Supervisor Call Selection Override? n
                                  Auto Reserve Agents: none
      Block Hang-up by Logged-in Auto-Answer Agents? n

 CALL MANAGEMENT SYSTEM
    REPORTING ADJUNCT RELEASE (determines protocol used by appl link)
                                       CMS (appl mis):
                                    AAPC/IQ (appl ccr):
                     ACD Login Identification Length: 0
                                BCMS/VuStats LoginIDs? n
                  BCMS/VuStats Measurement Interval: hour
          BCMS/VuStats Abandon Call Timer (seconds):
                     Validate BCMS/VuStats Login IDs? n
                           Clear VuStats Shift Data: on-login
                Remove Inactive BCMS/VuStats Agents? n
```

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid Entry	Usage
y	Include agents who are in After Call Work (ACW) mode in the Most-Idle Agent queue. This means that ACW is counted as idle time.
n	Exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill.

When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid Entry	Usage
all	Puts an agent on standby for all skills.
none	Agent is not on standby for any additional skills.
secondary-only	Puts an agent on standby only for secondary skills.

Block Hang-up by Logged-In Auto Answer Agents

The system displays the field only if you set the **EAS** field to y. The enhancement is available for agents using H.323 IP and DCP telephones, but does not apply and is not required for agents using BRI, analog, CallMaster, or SIPCC telephones.

Valid Entry	Usage
n	If an auto-answer agent goes on-hook when on an ACD call: • Communication Manager drops the active call and logs the agent out. • The reporting adjunct fails to record the call and track the logged-in agent.
y	You, as an administrator, can prevent an agent in the auto-answer mode from accidently logging out or dropping an active call. The agent can perform the following functions to drop calls: • Use the **release** button to disconnect the call and stay logged-in. • Use the **drop** button to release the last connected party. **Note:** If you set the field to y and the logged-in agent appearance is on-hook, a new call is not delivered automatically. The new call rings at the station and the agent must manually answer the call. If you have administered RONA for the skill, the call is redirected if the agent fails to answer within the administered number of rings.

Call Selection Measurement

Determines how Avaya Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

Valid Entry	Usage
current-wait-time	Selects the oldest call waiting for any of the agent's skills.
predicted-wait-time	Uses the time a call is predicted to wait in queue instead of the time the call has already waited. Available only if Business Advocate features are enabled for the system.

Copy ASAI UUI During Conference/Transfer

Enables or disables copying user-to-user (UUI) information during a conference call or during call transfers. Available only if ASAI Link Core Capabilities or Computer Telephony Adjunct Links are enabled for the system.

Note:

The system copies all UUI information, not just ASAI UUI. Copying occurs only during a human-initiated conference or transfer. Communication Manager does *not* copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Enables or disables the removal of an agent from the Most Idle Agent (MIA) queue for all available splits/skills/hunt groups when answering a call from any of these groups.

Service Level Maximizer Algorithm

Selects an alternative algorithm for selecting agents and delivering calls to maximize service level targets. Available only if Service Level Maximizer is enabled for the system.

Valid Entry	Usage
actual	The Actual Service Level (ASL) is determined as a percentage on a hunt group basis using the number of accepted calls in the current interval divided by the total calls in the current interval. A call is counted as accepted if it is answered within the target service level time period.
weighted	The Weighted Service Level (WSL) is based on a weighting calculation that uses the difference between the target time and the estimated wait time.

Service Level Supervisor Call Selection Override

Determines whether Avaya Communication Manager changes agent call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold. Available only if Expert Agent Selection (EAS) and the Business Advocate features are enabled for the system.

Valid Entry	Usage
y	Overrides the normal call handling preferences of a skill's assigned agents in this situation.
n	Normal call handling preferences are in effect when the skill exceeds its Level 1 threshold.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid Entry	Usage
y	Forces the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection.
n	Always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Enables or disables the transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEMS

REPORTING ADJUNCT RELEASE (determines protocol used by appl link)

CMS (appl mis)

Use this field to specify the release of the Call Management System (CMS) reporting adjunct. Management Information System (MIS) is a communication-interface link that connects CMS to Communication Manager.

Valid entry	Usage
R12	CMS Release 12 connects to Communication Manager through the mis1 link. CMS Release 12 is incompatible with Avaya IQ.
R13	CMS Release 13 connects to Communication Manager through the mis1 link. CMS Release 13 is incompatible with Avaya IQ.
R13.1	CMS Release 13.1 connects to Communication Manager through the mis1 link. CMS Release 13.1 and later is compatible with Avaya IQ.
R14.1	CMS Release 14.1 connects to Communication Manager through the mis1 link.
R15/R16	CMS Release 15 and 16 connects to Communication Manager through the mis1 link.
R16.1/R16.x	CMS Release 16.x and 17 connects to Communication Manager through the mis1 link.
blank	CMS does not connect to Communication Manager. This is the default value.

In the case of two CMS adjuncts, one CMS adjunct connects to Communication Manager through the mis1 link and the other CMS adjunct connects to Communication Manager through the mis2 link.

For more information about **CMS (appl mis)**, see *Administering Avaya Aura® Call Center Elite*.

IQ (appl ccr)

Expert Agent Selection (EAS) and Universal Call ID (UCID) must be enabled before a connection can be established with Avaya IQ.

Valid Entry	Usage
4.0 5.0 5.1	The release of Avaya IQ connected to the ccr1 link, and to the ccr2 link for a second Avaya IQ.
blank	Avaya IQ is disconnected. This is the default.

OTHER CALL MANAGEMENT SYSTEM FIELDS

ACD Login Identification Length

Available only if Expert Agent Selection (EAS) is disabled and BCMS/VuStats Login IDs are enabled for the system.

Valid Entry	Usage
1 to 9	The number of digits for an ACD agent login ID that identifies an ACD agent to theCall Management System. This number must equal the number of characters in the agent's login ID and cannot be 0.

BCMS/VuStats Abandon Call Timer (seconds)

Valid Entry	Usage
1 to 10 blank	The number of seconds before calls are abandoned. Calls with talk time that is less than this number and are not held, are tracked by the Basic Call Management System (BCMS) and displayed by VuStats as ABAND calls.

BCMS/VuStats LoginIDs

Enables or disables valid agent login IDs to monitor call activity by an agent. Basic Call Management System (BCMS) and VuStats login IDs are available in addition to Expert Agent Select (EAS) login IDs if EAS is enabled on the system. Both BCMS and CMS use the same login ID for an agent.

BCMS/VuStats Measurement Interval

Available only if BCMS (Basic) or VuStats is enabled for the system.

Valid Entry	Usage
half-hour hour	Selects measurement intervals for polling and reporting data. There are a maximum of 25 time slots available for measurement intervals. If hour is specified, an entire day of traffic information will be available for history reports. Otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval can be changed at any time, but remains until the current interval completes.

Clear VuStats Shift Data

Valid Entry	Usage
on-login	Clears shift data for an agent when the agent logs in.
at-midnight	Clears shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid Entry	Usage
y	Agents are removed from reports when they have no staff time during the previous seven days.
n	Agents remain on the report even if they have no staff time for any period of time.

Validate BCMS/VuStats Login IDs

Valid Entry	Usage
y	Only login-IDs that have been administered for the Basic Call Management System (BCMS) can be entered.
n	Any ACD login of the proper length can be entered.

Feature-related system parameters: page 13

```
change system-parameters features                              Page  13 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER MISCELLANEOUS
           Callr-info Display Timer (sec): 10
                        Clear Callr-info: next-call
       Allow Ringer-off with Auto-Answer? n

   Reporting for PC Non-Predictive Calls? n

           Agent/Caller Disconnect Tones? n

            Zip Tone Burst for Callmaster Endpoints: double

 ASAI
           Copy ASAI UUI During Conference/Transfer? n
      Call Classification After Answer Supervision? n
                                  Send UCID to ASAI? n
        For ASAI Send DTMF Tone to Call Originator? y
 Send Connect Event to ASAI For Announcement Answer? n
```

CALL CENTER MISCELLANEOUS

Allow Ringer-off with Auto-Answer

If activated, an agent can use the ringer-off feature button that prevents ringing on Expert Agent Selection (EAS) auto-answer calls.

Callr-info Display Timer (sec)

Valid Entries	Usage
3 to 60	Administer the timer to display the information which caller has dialed on IP (H.323) telephones. The default value is 10 secs.

Clear Callr-info

Specifies when the collected digits Callr-Info display is removed from the agent or station display.

Valid Entry	Usage
leave-ACW	Leaves the display up while the agent is in After Call Work (ACW) mode.
next-call	Clears the display when the next call is received. This is the default.
on-call-release	Clears the display on the second line of a two-line display telephone or the display of a one-line display telephone with the **Enhanced Callr-Info display for 1-line display phones** option active when the call is released. The call is released if the receiving call is disconnect or if the agent presses the **release** button.

Interruptible Aux Notification Timer (sec)

Valid Entry	Usage
1 to 9	Specifies the number of seconds the endpoint interruptible aux notifications, the flashing lamp, display, or tone, are on before an auto-in-interrupt or manual-in-interrupt agent is made available. This delay ensures that an agent is not immediately made available when he presses an interruptible aux button. Also, this delay provides a brief period to an agent already in interruptible Aux mode before that agent is made available automatically. Default is 3.

PC Non-Predictive Reports Skill

Administers a skill hunt group used for reporting associated with Proactive Contact non-predictive switch-classified calls on a per-system basis. Reports are generated as though the agent were in the ACD-OUT state. Available only if reporting for PC Non-Predictive Calls is enabled.

Valid Entry	Usage
1 to 99	Skill number for S8300 switches.
1 to 8000	Skill number for all other platforms

Reporting for PC Non-Predictive Calls

Activates or deactivates improved integration with Proactive Contact Outbound Calling for non switch-classified outbound calling. For example, this feature improves Call Management System tracking for switch-classified and non-switch classified (agent classified) outbound calls placed by the Proactive Contact soft dialer through ASAI. Default is n.

Send Connect Event to ASAI For Announcement Answer

Use this field to enable Communication Manager to send a Connected or Answered event to the calling party if the calling party is an ASAI application and the called party uses an announcement system. If you set this field to y, Communication Manager sends the Connected Answered event to the ASAI application when the announcement system at the called party answers the call. The announcement system answers the call by collecting the digits that the ASAI application provides. The system uses the **collect** command of the vector to collect the digits.

Note:

Communication Manager does not send a Drop event when the call gets disconnected.

Valid entry	Usage
n	Communication Manager does not send the Connected or Answered event to the ASAI application. This is the default value.
y	Communication Manager sends the Connected or Answered event to the ASAI application.

Note:

You should be careful while creating vectors that include announcements and other advanced features like route requests. The features can lead to:

- Call sequences in which Communication Manager sends a Connected event to the called party but do not send an Alerting event.
- Call sequences in which Communication Manager sends a Connected event to a calling party followed by an Alerting event to the same party but with different domains associated with the Connected event and the Alerting event.

Service Level Algorithm for SLM

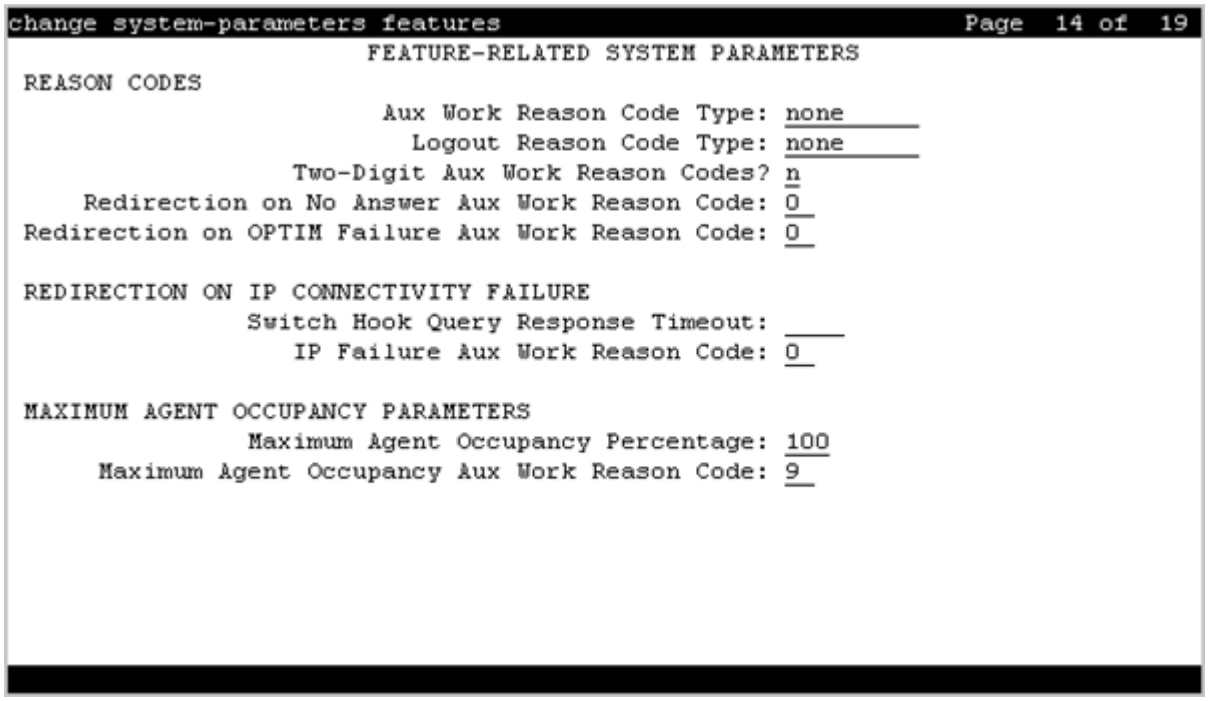
Administers the algorithm used to determine service levels for Service Level Maximizer (SLM). Available only if Expert Agent Selection (EAS) has been enabled for the system.

Valid Entry	Usage
actual	The Actual Service Level (ASL) based on an algorithm that is determined as a percentage on a hunt group (skill) basis using the number of calls answered within the target service level time period in the current interval, divided by the total calls in the interval. This is the default.
weighted	The Weighted Service Level (WSL) algorithm based on a weighting calculation that uses the difference between the target time and the estimated wait time.

Zip Tone Burst For Callmaster Endpoints

Valid entry	Usage
double	When the station set type is a Callmaster series, Communication Manager retains existing operation which applies the following to ACD agents: • two bursts of zip tone for auto-answer ACD calls • two burst of ICI tweedle-dee tone for non-ACD auto-answer calls This is the default value.
single	Communication Manager eliminates the 2nd burst of zip/ICI tone reducing time for agent to start conversation with the caller and possibility of the agent and the caller hear open mike background noise between the first and second tones. Use with a Callmaster station type when the agent can always hear enough of the single burst auto-answer to recognize that a call is being delivered.

Feature-related system parameters: page 14



REASON CODES

Aux Work Reason Code Type

Valid Entry	Usage
none	An agent does not enter a Reason Code when entering AUX work.
requested	An agent can enter a Reason Code when entering AUX mode but is not forced to do so. Available only if Reason Codes and EAS are enabled for the system.
forced	An agent is forced to enter a Reason Code when entering AUX mode. Available only if Reason Codes and EAS are enabled for the system.

Logout Reason Code Type

Valid Entry	Usage
none	An agent does not enter a Reason Code when logging out.
requested	An agent can enter a Reason Code when logging out but do not want to force the agent to do so. Available only if Reason Codes and EAS are enabled for the system.
forced	An agent is forced to enter a Reason Code when logging out. Available only if Reason Codes and EAS are enabled for the system.

Redirection on No Answer Aux Work Reason Code

Redirection on No Answer (RONA) redirects a ringing ACD skill or direct agent call after an administered number of rings. When RONA is invoked for a call, the system places an agent in the Aux work mode, and the agent is unavailable to receive calls from other splits and skills. This field is used to define the Reason Code for reporting, instead of the default of 0, when the state of the agent is changed to Aux due to RONA.

Valid Entry	Usage
0 to 99	If the **Two-Digit Aux Work Reason Code** field is set to y.
0 to 99	If the **Two Digit Aux Work Reason Code** field is set to n.

Redirection on OPTIM Failure Aux Work Reason Code

Redirection on OPTIM Failure (ROOF) applies only to ACD agents using SIP hard or soft phones administered as Off-PBX Telephone Integration and Mobility (OPTIM) endpoints. ROOF provides redirection of calls back into queue or to the specified VDN. The agent is not aware that the line is in the Aux work mode during an IP connectivity failure. If connectivity is restored during the IP network TCP retry period, the lamp will indicate that the line is in the Aux work mode. The value in the field is used to define the Reason Code for reporting, instead of the default of 0, when the state of the agent is changed to Aux due to ROOF.

Valid Entry	Usage
0 to 99	If the **Two-Digit Aux Work Reason Code** field is set to y.
0 to 99	If the **Two Digit Aux Work Reason Code** field is set to n.

Two-Digit Aux Work Reason Codes

Enables or disables two-digit reason codes for agent state changes for AUX Work.

REDIRECTION ON IP CONNECTIVITY FAILURE

Auto-answer IP Failure AUX Reason Code

Valid Entry	Usage
0 to 99	The reason code assigned for auto-answer IP failure, as the reason the agent was put into AUX Work..

Switch Hook Query Response Timeout

Valid Entry	Usage
500 to 5000 (msec)	The time that call processing waits for a response from the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered.
blank	ROIF is inactive.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA).

When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available.

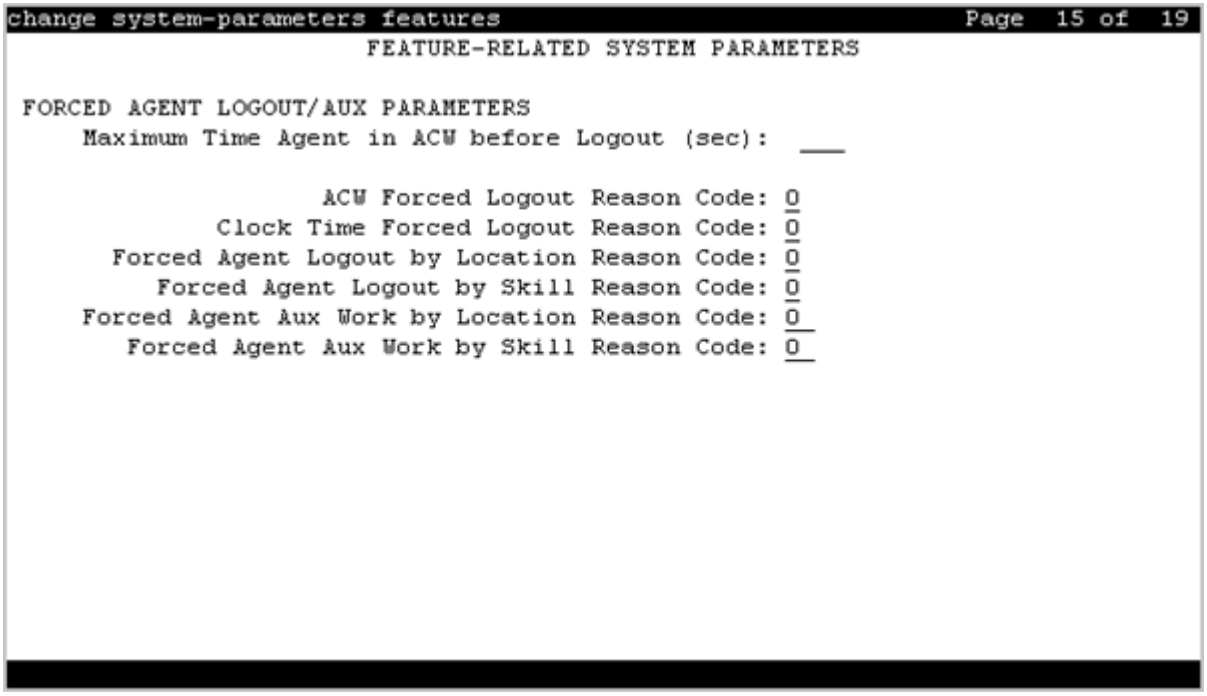
Maximum Agent Occupancy AUX Reason Code

Valid Entry	Usage
0 to 99	A reason code value. Do not use reason code 0. Default is 9.

Maximum Agent Occupancy Percentage

Valid Entry	Usage
0 to 100	The maximum percentage of time an agent can be taking calls. This time is based on Maximum Agent Occupancy (MAO) calculations. Default is 100.

Feature-related system parameters: page 15



FORCED AGENT LOGOUT PARAMETERS

ACW Forced Logout Reason Code

Available only if the Call Center Release is 3.0 or later and Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
0 to 9	The reason for logging out the agent due to time-out in After Call Work (ACW).

Clock Time Forced Logout Reason Code

Valid Entry	Usage
0 to 9	The reason code for the Forced Agent Logout by Clock Time feature used by administrators to set a specific time when the system automatically logs out Expert Agent Selection (EAS) agents.

Maximum Time Agent in ACW before Logout (sec.)

Available only for Call Center Release 3.0 or later and if Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
30 to 9999	A system-wide maximum time an agent can be in After Call Work (ACW). When this timer expires, the agent is logged out.
blank	There is no time-out. This is the default.

Feature-related system parameters: page 16

```
change system-parameters features                              Page  16 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

SPECIAL TONE
                                    Special Dial Tone? n
           Special Dial Tone for Digital/IP Stations: none

REDIRECTION NOTIFICATION
                              Display Notification for Do Not Disturb? n
                              Display Notification for Send All Calls? n
                                Display Notification for Call Forward? n
                        Display Notification for Enhanced Call Forward? n
                             Display Notification for a locked Station? n
           Display Notification for Limit Number of Concurrent Calls? n
                              Display Notification for Posted Messages? n
                                    Scroll Status messages Timer(sec.): __




Chained Call Forwarding? n
```

SPECIAL TONE

Special Dial Tone

Enables or disables an audible tone indicating that the station is locked.

Related Links

Special Dial Tone on page 294

Special Dial Tone for Digital/IP Stations

Valid Entry	Usage
all none non-display	System plays special dial tone for digital or IP stations, if the Limit Number of Concurrent Calls, Station Lock, or Call Forward feature is active.

REDIRECTION NOTIFICATION

Chained Call Forwarding

Enables or disables Chained Call Forwarding. With Chained Call Forwarding, calls can be forwarded to as many as 10 forwarded-to stations using a preset forwarding path.

Display Notification Enhanced Call Forward

Enables or disables display notification for Enhanced Call Forward.

Display Notification for a locked Station

Enables or disables display notification for a locked station.

Display Notification for Call Forward

Enables or disables display notification for Call Forward.

Display Notification for Do Not Disturb

Enables or disables display notification for Do Not Disturb.

Display Notification for Limit Number of Concurrent Calls

Enables or disables display notification for Limit Number of Concurrent Calls.

Display Notification for Posted Messages

Enables or disables display notification for posted messages.

Display Notification for Send All Calls

Enables or disables display notification for Send All Calls.

Scroll Status messages Timer (sec.)

Valid Entry	Usage
5 to 10	The time in seconds for displaying scroll status messages. The status information shows the feature that has the highest priority and is enabled. The features in order of decreasing priority are: • Do Not Disturb • Send All Calls • Call Forward • Posted Messages • Limit Number of Concurrent Calls (LNCC) • Station Lock
blank	Deactivates the scrolling. This is the default.

Feature-related system parameters: page 17

```
change system-parameters features                              Page  17 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS
                       Automatic Exclusion by COS? n




                              Recall Rotary Digit: 2

         Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1: _____    2: _____    3: _____    4: _____    5: _____



RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                                Re-try? n
     T2 (Backward Signal) Activation Timer (secs): 20
```

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Valid Entry	Usage
y	Enables automatic exclusion by a class of service when a user goes off-hook on a station with an assigned **Exclusion** button. This works only for stations on the local server running Communication Manager.
n	Exclusion operates normally.

Automatic Exclusion Coverage/Hold

Available only when Automatic Exclusion by COS is enabled.

Valid Entry	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Available only Automatic Exclusion by COS is enabled.

Valid Entry	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Valid Entry	Usage
3 to 30	The length of time in 3 second increments that the call information remains on display after the call is terminated.

Password to Change COR by FAC

If this field contains a value, a password option is required. Accepts from four to eight digits.

Available only if **Change COR by FAC** is enabled for the system.

Recall Rotary Digit

Valid Entry	Usage
0 to 9	The digit used for rotary telephones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary telephones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary. This should be a number that is not the first digit in normal dialing patterns.

WIRELESS PARAMETERS

Radio Controllers with Download Server Permission

The port location of the circuit pack containing the radio controllers with download server permission.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier.
0 to 20	Fourth and fifth characters are the slot number.

IP PARAMETERS

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

IP Audio Hairpinning

If enabled, IP endpoints connect through the IP circuit pack in the server in IP format to bypass the Communication Manager TDM bus.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

Enables or disables the resending of address information on outgoing Russian MFP trunks. Specifically, the server running Communication Manager resends Russian MFP calling party number and dialed number information to the local telephone company central office (CO). The server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the

CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

T2 (Backward Signal) Activation Timer (secs)

Valid Entry	Usage
5 to 20	The number of seconds Communication Manager waits to receive confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks.

Feature-related system parameters: page 18

```
change system-parameters features                              Page  18 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
       Invalid Number Dialed Intercept Treatment: tone
                     Invalid Number Dialed Display:
    Restricted Number Dialed Intercept Treatment: tone
                  Restricted Number Dialed Display:
   Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
   Whisper Page Tone Given To: all

6400/8400/2420J LINE APPEARANCE LED SETTINGS
                     Station Putting Call On Hold: green  wink
                      Station When Call is Active: steady
         Other Stations When Call Is Put On Hold: green  wink
              Other Stations When Call Is Active: green
                                        Ringing: green  flash
                                           Idle: steady

                                Pickup On Transfer? y
```

INTERCEPT TREATMENT PARAMETERS

Intercept Treatment on Failed Trunk Transfers

Valid Entry	Usage
y	Provides intercept treatment to calls failing trunk transfers.
n	Drops calls failing trunk transfers.

Invalid Number Dialed Display

Used to display a name in either Latin or Asian characters for an invalid number calling in. This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters that can be displayed. A value of NAME2 directs the system to use the UTF-8 table of names that contains non-ASCII characters suitable for Asian language names. Accepts up to 15 alphanumeric characters.

Note:

The system displays the message administered for the **Invalid Number Dialed Display** field on the telephone only if the **Invalid Number Dialed Intercept Treatment** field is set to `announcement`.

Invalid Number Dialed Intercept Treatment

The type of intercept treatment the end-user hears after dialing an invalid number.

Valid Entry	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Requires entering the extension number for the announcement.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Restricted Number Dialed Display

The string of alphanumeric characters assigned for calls that are denied because of COS, COR, or FRL restrictions. This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters. A value of NAME2 directs the system to use the UTF-8 table of names that contains non-ASCII characters suitable for Asian language names. Accepts up to 15 alphanumeric characters.

Note:

The system displays the message administered for the **Restricted Number Dialed Display** field on the telephone only if the **Restricted Number Dialed Intercept Treatment** field is set to `announcement`.

Restricted Number Dialed Intercept Treatment

The type of intercept treatment the caller hears after dialing a number restricted from them due to COS, COR, or FRL restrictions.

Valid Entry	Usage
tone	Provides intercept tone. This is the default.
announcement	Provides a recorded announcement. You select and record the message. Requires entering the extension number for the announcement.

WHISPER PAGE

Whisper Page Tone Given To

Determines who should hear a Whisper Page beep tone.

Valid Entry	Usage
all	All parties hear the whisper page beep tone.
paged	The whisper page feature sends a beep tone only to the paging and the paged party.

6400/8400/2420J LINE APPEARANCE LED SETTINGS

Warning:

The following fields change only the LED operation for 84*xx* and 64*xx* model telephones. When the LED operation is changed using any of these fields, then IP Agent and IP Softphone using a station type of 84*xx* or 64*xx* does not work. For station types other than 84*xx* or 64*xx*, a change to the LEDs using these fields does not affect either IP Agent or IP Softphone.

Note:

The system generates a warning if the default values of the LED Settings field are changed. The warning message states

```
WARNING: Avaya Softphone will not operate correctly if this value is changed
```

. This warning message displays for Avaya Communication Manager 3.1 or higher.

Display Information With Bridged Call

Controls whether or not name and number for a bridged call are displayed on the telephone of the called party. This field does *not* control the content of the display.

Idle

Note:

This field applies only to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs.

Valid Entry	Usage
steady	The LED flash rate for an idle station. Default is steady.
off	The correct value for the Japanese environment is off.

Other Stations When Call Is Active

Controls a DCP bridged appearance LED for those non-active parties with a bridged appearance that is active.

Note:

This field applies only to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs.

Valid Entry	Usage
green	The LED color. Default is green.
red	Red is the correct value in the Japanese environment.

Other Stations When Call Is Put On-Hold

Controls LED options for the other stations with a Bridged Appearance that have been placed on hold, but the user of this station has not pressed the hold button.

Note:

This field is for a DCP bridged appearance LED color and flash rate when a call on a bridged appearance is put on hold by another party on the DCP bridged appearance. Additionally, this field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values red and flash for this field.

Valid Entry	Usage
green red	The color of the LED. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate for the LED. Default is wink.

Pickup on Transfer

Valid Entry	Usage
y	A station with a bridged appearance can pick up a call on hold because of a transfer.
n	Another station with bridged appearances cannot pick up a call on hold because of a transfer.

Ringing

Controls the LED color and flash rate while a call is ringing.

Note:

This field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values red and wink for this field.

Valid Entry	Usage
green red	The LED color. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate. Default is flash.

Station Putting Call On-Hold

Controls the LED color and flash rate on the 8400 and 6400 series telephones for a call held on a Primary or Bridged Appearance.

Valid Entry	Usage
green red	The color of the LED. The LED for the color not selected is turned OFF. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate for a call on hold. Default is wink.

Station When Call is Active

Controls the red LED on the 8400 and 6400 series telephones, for a station active on a call.

Valid Entry	Usage
steady	Communication Manager controls the red LED. This is the default.
off	The red LED is always OFF.

Feature-related system parameters: page 19

```
change system-parameters features                              Page  19 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
                    Direct IP-IP Audio Connections? y
                              IP Audio Hairpinning? y
                           Synchronization over IP? n
               SDP Capability Negotiation for SRTP? y
                     SIP Endpoint Managed Transfer? y

CALL PICKUP
  Maximum Number of Digits for Directed Group Call Pickup: 4
                   Call Pickup on Intercom Calls? y     Call Pickup Alerting? y
     Temporary Bridged Appearance on Call Pickup? y     Directed Call Pickup? y
                        Extended Group Call Pickup: none
                   Enhanced Call Pickup Alerting? n


                         Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                  PIN Checking for Private Calls? n
```

IP PARAMETERS

Allow Bridge DM Answer

Use this field to enable the Bridge call answer as Direct Media feature. To enable this feature, set the **SIP Endpoint Managed Transfer** and **Allow Bridge DM Answer** field to `y` .

Valid entry	Usage
y	The calling party can have direct media path with the bridge call appearance of the called party station if all the involved stations support the Enhanced SIP Signaling feature.
n	This is the default value.

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

IP Audio Hairpinning

If enabled, IP endpoints connect through the IP circuit pack in the server in IP format to bypass the Communication Manager TDM bus.

Synchronization over IP

Enables or disables the Clock Synchronization over IP (CSoIP) feature to provide IP sync to a group of members.

Valid entries	Usage
y	Enables the CSoIP feature.
n	Disables the CSoIP feature. This is the default value.

SDP Capability Negotiation for SRTP

The system displays SDP Capability Negotiation for SRTP when the **Media Encryption Over IP** field is set to y. Enables or disables SIP SRTP enhancements with Direct Media support in Communication Manager using Session Description Protocol (SDP) capability negotiation.

SIP Endpoint Managed Transfer

Use this field to enable the **SIP Endpoint Managed Transfer** feature. Before enabling the feature, you must ensure that the **Network Call Redirection** field is disabled for the SIP trunk group between Communication Manager and Session Manager.

Note:

The **SIP Endpoint Managed Transfer** field only controls the SIP transfer changes made in release 6.2. The field does not control some related Communication Manager SIP conference changes.

For more information on the **SIP Endpoint Managed Transfer** feature and on the related SIP conference changes, see *Avaya Aura Communication Manager Feature Description and Implementation*.

CALL PICKUP

A pickup group is a collection, or list, of individual extensions that have permission to answer calls originally dialed towards a different member of the group.

Audible Notification

Valid Entry	Usage
1 to 15 seconds	Set the audio notification timer. When the timer is set and a call arrives to a group member, the other members of the group are informed with an audio notification when the timer expires.

Call Pickup Alerting

Enables or disables Call Pickup Alerting on a system-wide basis. Call Pickup Alerting provides pickup group members with a visual indication on the **Call Pickup** status lamp for calls eligible to be answered using Call Pickup.

Call Pickup on Intercom Calls

System-wide users can attend or drop an intercom call using the Call Pickup or Directed Call Pickup features.

Directed Call Pickup

Activates or deactivates the use of Directed Call Pickup, where users can specify what ringing telephone they want to answer.

Enhanced Call Pickup Alerting

Valid entries	Usage
y	Apply the Enhanced Call Pickup Delay Timer.
n	Silently flash the pickup button.

Enhanced Call Pickup Delay Timer (sec.) Display

Valid Entry	Usage
1 to 15	When a call arrives at one of the group members the other members are informed with an audio notification. Set the timer in seconds to delay the audio notification. When the timer expires the audio notification is sent.

Extended Group Call Pickup

Selects how call pickup groups can answer calls directed to another call pickup group.

Valid Entry	Usage
flexible	A one-to-*n* pickup group-to-extended pickup group mapping.
simple	A one-to-one pickup group-to-extended pickup group mapping.
none	Extended group call pickup is prohibited.

Keep Bridged Information on Multiline Displays During Calls

If this field is enabled, for Multiline Display sets (1603, 1608, 1616, 4610, 4620, 4620SIP, 4620SIPCC, 4622, 4630, 9620, 9630 and 9650), the system displays the incoming caller information for the duration of the call. The system clears the caller information on the display for the user who puts the call on hold but retains the information on the display for the other users on the bridge. For outgoing calls, if the user on the bridge puts the call on hold, the display for the other user on the bridge is populated with the caller information. The default is n.

Maximum Number of Digits for Directed Group Call Pickup

Valid Entry	Usage
1 to 4	The maximum number of digits accepted for the pickup group number. The pickup group number is complete when it is followed by a # symbol. Default is 4.

PIN Checking for Private Calls

Enables or disables the PIN Checking for Private Calls feature that restricts users from making internal or external private calls by forcing them to enter a PIN code after dialing a PIN Feature Access Code (FAC).

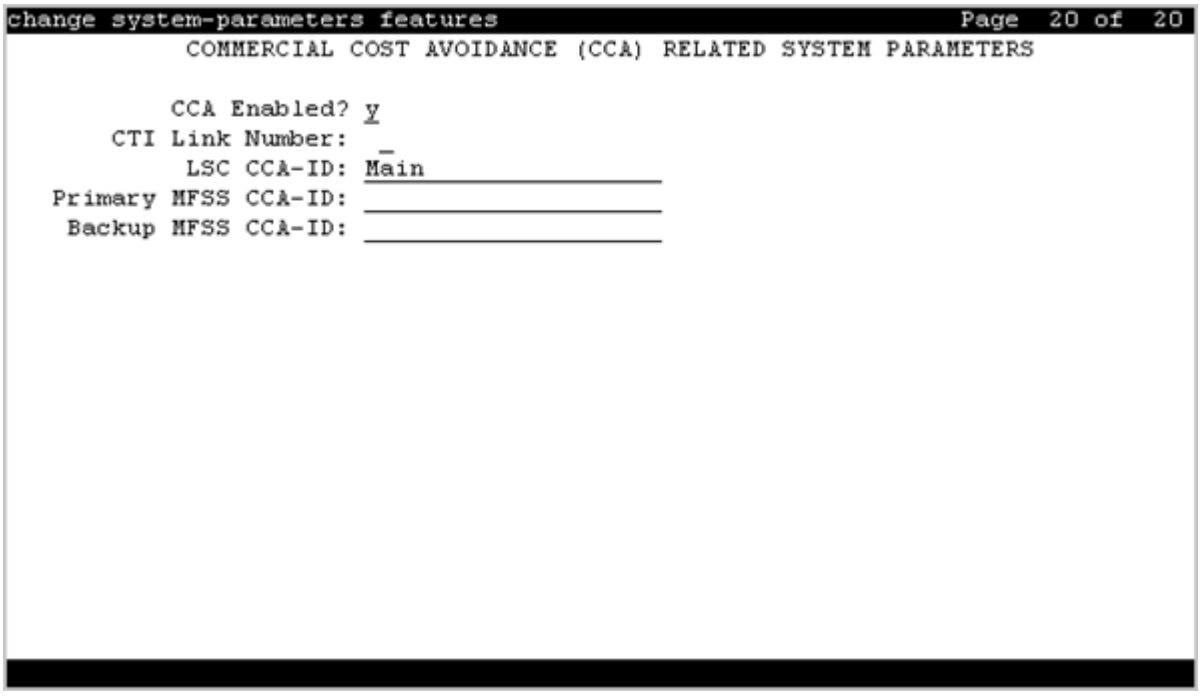
Related Links

PIN Checking for Private Calls Using AAR Access Code on page 254
PIN Checking for Private Calls Using ARS Access Code on page 254
PIN Checking for Private Calls Access Code on page 254

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter y to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter n to prevent the temporary bridged appearance of calls answered with these features.

Commercial Cost Avoidance (CCA) Related System Parameters



Backup MFSS CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. Enter a string without any white space.

CCA Enabled

Use this field to enable **Cost Commercial Avoidance (CCA)**.

Valid entry	Usage
y	The system displays the CCA-related fields.
n	The system does not display the CCA-related fields. The default value is n.

CTI Link Number

The CTI link number establishes a connection between Communication Manager and AES applications. Enter a CTI Link number of type ASAI-IP or ADJ-IP that is not used in any vector.

LSC CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. The default value of this field is the value populated in the **Call Control Agent Identification (CCA-ID)** field of MLPP. Though the value in the **LSC CCA-ID** field is from the **Call Control Agent Identification (CCA-ID)** field, you can change the value of the **LSC CCA-ID** field.

Note:

You can confirm the value of **Call Control Agent Identification (CCA-ID)** field on page 2 of the change system-parameters mlpp screen.

Primary MFSS CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. Enter a string without any white space.

Chapter 54: Firmware Station Download

This screen downloads firmware to multiple stations of the same telephone type, either 2420 or 2410 DCP telephones. Downloads firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

Example command: `change firmware station-download`

Beginning Station

The first extension number in the range of telephones used to download the firmware. Up to 1000 stations can be included in a scheduled download. Accepts up to 13 digits.

Continue Daily Until Completed

Enables or disables the execution of a firmware download each day at the scheduled time until all specified telephones have received the firmware.

Download Set Type

Valid Entry	Usage
2410 DCP 2420 DCP	The set type of DCP telephones to which firmware is to be downloaded.

Download Station Type

Type of station to which the firmware is to be downloaded.

Ending Station

The last extension number in the range of telephones used to download firmware. Up to 1000 stations can be included in a scheduled download. Accepts up to eight digits.

Schedule Download

Enables or disables a request to schedule a time for a firmware download to multiple DCP stations.

Source File

The name of the file used to retrieve the firmware download. Accepts up to 32 alphanumeric characters.

Related Links

File to Retrieve on page 845

Start Date/Time

Available only if **Schedule Download** is enabled.

Valid Entry	Usage
mm, dd, yyyy; hh, mm	The month, day, year, and time for the firmware download to begin.

Related Links

Schedule Download on page 347

Stop Date/Time

Available only if **Schedule Download** is enabled.

Valid Entry	Usage
mm, dd, yyyy; hh, mm	The month, day, year, and time for the firmware download to end.

Related Links

Schedule Download on page 347

Chapter 55: Group Paging Using Speakerphone

Assigns digital speakerphones to a paging group. Users can page all the telephones in the group simultaneously by dialing the group extension.

Example command: `change group-page n`, where *n* is the assigned group number.

Alert

Set this field to `y` for telephones that require an alert message to ring for an inbound call from a group page number. For example, Spectralink wireless telephones. Communication Manager does not activate the speakerphone of the Spectralink wireless telephones for a group page number. Therefore, the users do not hear the call ring from a group page number until the users manually answer the call.

Note:

Most of the Avaya telephones have the auto-answer capability, so if you set the **Alert** field to `y`, an Avaya telephone does not ring because the group paging initiates immediately.

Value	Usage
y	Telephones, such as Spectralink wireless, receive an alert to ring for an inbound call from a group page number.
n	Telephones, such as Spectralink wireless, do not receive an alert to ring for an inbound call from a group page number. This is a default value.

ASAI

Value	Usage
y	Enables ASAI events for paging groups. Enabling this functionality creates large spikes in the messaging traffic to the application.
n	ASAI events are not forwarded to the application for any call that has a paging group as parties on the call.

COR

Valid Entry	Usage
0 to 995	A class of restriction (COR). To page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Assigns a telephone extension to the group.

Group Extension

The extension users dial to page the members of this group.

Group Name

A name for the group that is informative to users. The callers' telephone displays when the caller pages the group. Accepts up to 27 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Group Number

The identifying number the server running Communication Manager assigns to the group when it is created.

Group Timeout (secs)

Valid Entry	Usage
0 to 600	Administer a timeout in seconds for the group page. After timeout, the paging party will be disconnected. The default value is 0 for no timeout.

Name

The name assigned to each extension in the group.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

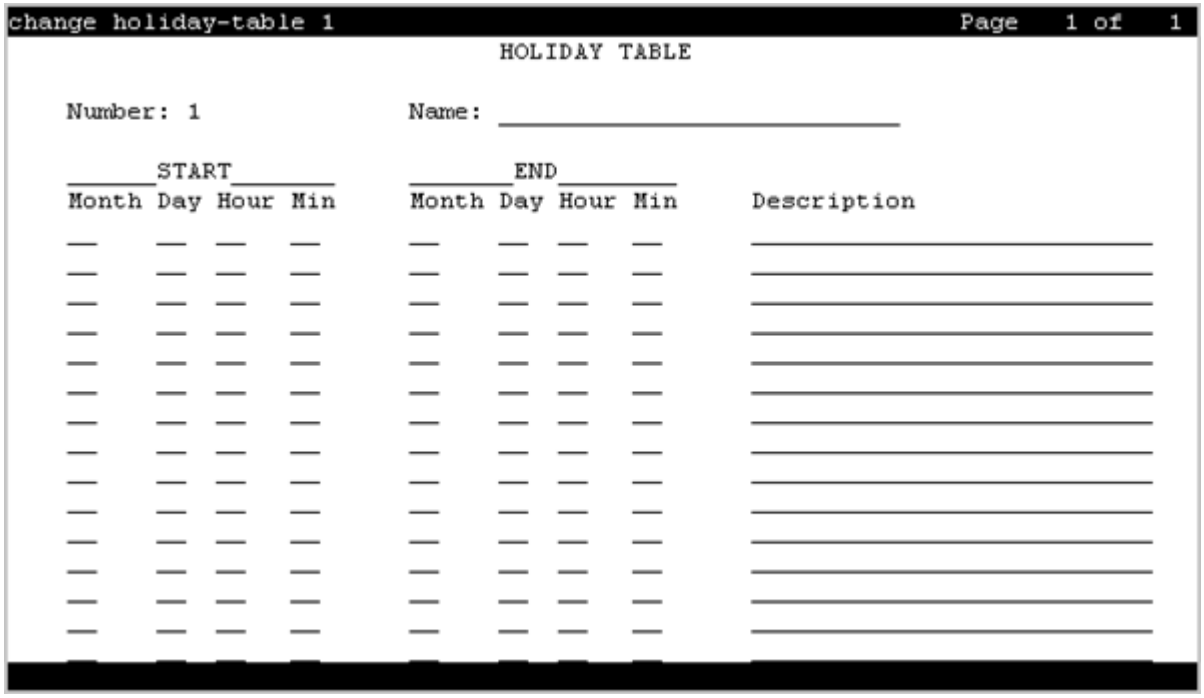
TN

The tenant number for this paging group. With tenant number, group paging can be partitioned by tenant.

Chapter 56: Holiday Table

Defines individual holidays or holiday ranges.

Example command: `change holiday-table n`, where *n* is the holiday table number.



Description

A phrase that describes the holiday. Accepts up to 27 characters.

End Day

Valid Entry	Usage
1 to 31	The ending day of the holiday.

Comments on this document? infodev@avaya.com

End Hour

Valid Entry	Usage
0 to 23	The ending hour of the holiday using a 24-hour clock.

End Min

Valid Entry	Usage
0 to 59	The ending minute of the holiday.

End Month

Valid Entry	Usage
1 to 12	The ending month of the holiday.

Name

The name of the holiday table. Accepts up to 27 characters.

Number

Valid Entry	Usage
1 to 10	The holiday table number.

Start Day

Valid Entry	Usage
1 to 31	The starting day of the holiday.

Start Hour

Valid Entry	Usage
0 to 23	The starting hour of the holiday using a 24-hour clock.

Start Min

Valid Entry	Usage
0 to 59	The starting minute of the holiday.

Start Month

Valid Entry	Usage
1 to 12	The starting month of the holiday.

Chapter 57: Hospitality

Implements the system parameters associated with the hospitality features. Available only if Hospitality features are enabled for the system.

Example command: `change system-parameters hospitality`

Related Links

Hospitality (Basic) on page 800
Hospitality (G3V3 Enhancements) on page 800

Hospitality: page 1

Client Room Coverage Path Configuration

Indicates the server and the Property Management System (PMS) exchange coverage path information for guest stations.

Valid Entry	Usage
act-nopms	The message is acknowledged (`MESSAGE ACK`), but no action is taken.
act-pms	The server and PMS exchange and accept coverage path information. This field does not apply to normal PMS Protocol mode.

Controlled Restrictions Configuration

Indicates whether controlled restriction information is exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (`MESSAGE ACK`), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Default Coverage Path for Client Rooms

Applies only to stations with a client room class of service in the occupied mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with client room class of service.

Valid Entry	Usage
1 to 9999 blank	The coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid Entry	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

Housekeeper Information Configuration

Indicates whether housekeeper information is exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (`MESSAGE ACK`), but no action is taken.
act-pms	The server and PMS exchange and accept housekeeper information.

Journal/Schedule Endpoint

Valid Entry	Usage
Valid data extension number	A valid data extension number that is assigned to the data module connected to the Journal/Schedule printer. Cannot be a VDN extension. This extension can be the same as the PMS/ Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	The printer is connected over a TCP/IP link, and this link is administered with a **PMS_LOG** service type for IP Services.
PMS_JOURNAL	The printer is connected over a TCP/IP link, and this link is administered with a **PMS_JOURNAL** service type for IP Services.

Related Links

Service Type on page 474

Message Waiting Configuration

Indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (`MESSAGE ACK`), but no action is taken.
act-pms	Message waiting is active on the server and information between the PMS and server is transmitted.

Number of Housekeeper ID Digits

Valid Entry	Usage
0 to 6	The number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

Valid Entry	Usage
Valid extension	The data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	The PMS is connected over a TCP/IP link. This link is administered with PMS service type for IP Services.

Related Links

Service Type on page 474

PMS LINK PARAMETERS

ASCII mode

Enables or disables ASCII-only mode used for the PMS message set. Available only with a transparent PMS protocol mode.

PMS Link Maximum Retransmission Requests

Valid Entry	Usage
1 to 5	The number of times that the PMS can request acknowledgment for a message sent to the server.

PMS Link Maximum Retransmissions

Valid Entry	Usage
1 to 5	The number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

Milliseconds Before PMS Link Acknowledgment Timeout

Regulates how quickly the system responds to a message from the PMS, also known as “pace timing”. This value is also used as the inquiry message (ENQ) time-out value. Should be kept as short as possible.

Valid Entry	Usage
100 to 1500	The time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Endpoint

Valid Entry	Usage
Valid extension	The data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	The PMS is connected over a TCP/IP link. This link is administered with PMS service type for IP Services.

Related Links

Service Type on page 475

PMS Protocol Mode

Valid Entry	Usage
normal transparent	Indicates the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.

Seconds Before PMS Link Idle Timeout

Valid Entry	Usage
5 to 20	The idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Take Down Link for Lost Messages

Enables or disables taking down the PMS link if messages are getting lost. The PMS error log should be monitored if disabled.

Hospitality: page 2

Announcement Ports

Indicates the equipment location of two ports on the voice synthesizer circuit pack. Available only with a voice-synthesis announcement type.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number.
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Related Links

Announcement Type on page 359

Announcement Type

Indicates the type of automatic wake up announcement the hotel guest receives.

Valid Entry	Usage
external	Applicable when using an announcement adjunct. Requires entry of the circuit connection to the external announcement equipment.
integrated	Applicable when using the TN750B or TN750C announcement circuit pack. Requires entry of an extension for the integrated announcement.
mult-integ	Multi-integrated is applicable when using the TN750B or TN750C announcement circuit pack. Automatic Wakeup can use integrated announcement circuit packs to

Table continues…

Valid Entry	Usage
	play any one of multiple announcements to different extensions during a wake up call. Requires entry of an announcement extension.
voice-synthesis	A voice synthesis message is heard during the wake up announcement. Requires entry of port location information for the voice synthesizer circuit pack.
music-on-hold	Uses the Music-on-Hold feature to provide the wake up announcement.
silence	Silence is heard during the wake up announcement.

Related Links

Announcement Ports on page 359
Auxiliary Board for Announcement on page 360
Integrated Announcement Extension on page 362
Default Announcement Extension on page 361

Automatic Selection of DID Numbers

Enables or disables the Automatic Selection of DID Numbers for Guest Rooms feature. This feature assigns a two- to five-digit extension from a predetermined list of numbers to a hotel room telephone number that is disassociated with the room number.

Auxiliary Board for Announcement

The equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Available only for an external announcement type.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number.
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Related Links

Announcement Type on page 359
Integrated Announcement Extension on page 362
Default Announcement Extension on page 361

Custom Selection of VIP DID Numbers

The selection or removal of a DID number assigned to a room when a guest checks in. Available only if **Automatic Selection of DID Numbers** is enabled.

Related Links

Automatic Selection of DID Numbers on page 360

Daily Wakeup

Each extension permission can request daily wake up calls.

Default Announcement Extension

The default wake up announcement extension when using the integrated announcement circuit pack.

Available only with a multi-integrated announcement type.

Related Links

Announcement Type on page 359
Auxiliary Board for Announcement on page 360
Integrated Announcement Extension on page 362

Digit to Insert/Delete

Valid Entry	Usage
0 to 9	The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. To accommodate PMS devices that are based on room number and not extension. This leading digit can be deleted on messages from Avaya Communication Manager to the PMS, and then inserted back on messages from the PMS to Communication Manager. **Note:** The PMS interface supports three-, four-, or five-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, do not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Display Room Information in Call Display

Indicates the type of guest room information displayed on telephone displays.

Valid Entry	Usage
y	Telephones display the name and room number. The extension number and room number are not always the same number.
n	Telephones display the name and extension number.

Dual Wakeup

Allows or disallows each extension permission to request two wake up calls within one 24-hour period.

Extension to Receive Failed Wakeup LWC Messages

An extension that indicates where unsuccessful wake up LWC messages are stored. This is usually administered to an unassigned extension or to the attendant (attd). This extension cannot be a VDN extension. In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wake up calls.

Integrated Announcement Extension

The wake up announcement extension when using the integrated announcement circuit pack.

Available only for integrated announcement types.

Related Links

Announcement Type on page 359
Auxiliary Board for Announcement on page 360
Default Announcement Extension on page 361

Length of Time to Remain Connected to Announcement

Valid Entry	Usage
0 to 300	The length of time in seconds that a hotel guest is connected to an announcement. Applies only after the guest has heard the announcement completely one time, but continues to listen.

Number of Digits from PMS

The number of digits being sent from the PMS to the server to identify room numbers.

Valid Entry	Usage
1 to 4	For normal mode
1 to 5	For transparent or ASCII mode
blank	For mixed numbering in the server.

Number of Digits in PMS Coverage Path

Valid Entry	Usage
3 or 4	The number of digits in the coverage path.

PMS Sends Prefix

Enables or disables PMS Sends Prefix to Indicate if the PMS sends a prefix digit to the server as part of the room numbering plan.

Room Activated Wakeup with Tones

Enables or disables the activation of wake up calls with tones. Wake up calls can be activated with tones that prompt users for the time they want to wake up. This allows room activated wake up calls without the use of a speech synthesizer or a display telephone.

Routing Extension on Unavailable Voice Synthesis

Valid Entry	Usage
Assigned extension	A call is placed to this extension or to the attendant if a voice synthesis port is unavailable during voice synthesis entry of wakeup requests. This extension cannot be a VDN extension.
attd	An attendant group code.

Time of Scheduled Emergency Access Summary Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Emergency Access Summary Report gets printed on the Journal/ Schedule printer.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

Time of Scheduled Wakeup Activity Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Wakeup Activity Report gets printed on the Journal/ Schedule Printer. This report summarizes the wake up activity for each extension that had wake up activity for the past 24 hours.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

Time of Scheduled Wakeup Summary Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Wakeup Summary Report gets printed on the Journal/ Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wake up calls and a list of extensions to which wake up calls were attempted but were left incomplete during the hour.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

VIP Wakeup

Allows or disallows permission for each extension to request VIP wake up calls.

VIP Wakeups Per 5 Minutes

Available only if **VIP Wakeup** is enabled for each extension.

Valid Entry	Usage
1 to 50	The number of VIP Wakeup calls allowed in a 5-minute interval.

Related Links

VIP Wakeup on page 364

Hospitality: page 3

ROOM STATES

Definition for Rooms in State 1 through 6

A definition for each room status. These definitions are only for **Attendant Room Status**. Accepts up to 30 characters.

Example

State 1 could be “clean, ready to use” and state 2 could be “occupied, needs cleaning”.

HOSPITALITY FEATURES

Cancel Do-Not-Disturb for Wakeup Calls

Activates and deactivates whether a wakeup call will cancel Do-not-disturb.

Suite Check-in

Allows or disallows attendants permission to have the system automatically check-in several related extensions with one check-in command.

Chapter 58: Hunt Group

With Hunt groups, users or agents can answer calls at a predefined group of telephones or devices.

Use the Hunt Group screen to create a hunt group, identified by a hunt group number, and to assign hunt group member users by their extension numbers. This screen can also be used to implement associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing.

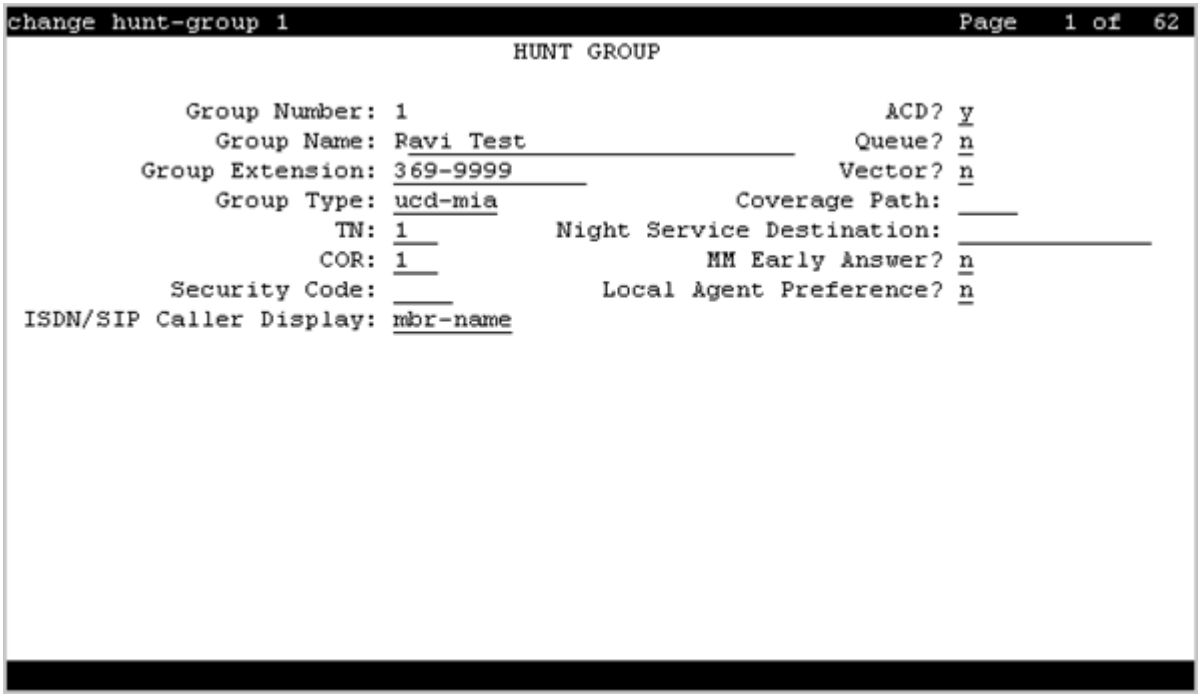
When a call comes into a hunt group, the system checks for the busy or idle status of extension numbers in the hunt group when answering. A Uniform Call Distribution (UCD) type hunt group selects the “most idle” extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the “most idle” agent or the “least occupied” agent with the highest skill level for the call’s skill.

Note:

Vector controlled splits/skills can be called directly through the split/skill extension instead of calling a VDN mapped to a vector that terminates the call to a vector controlled split/skill. However, the calls do not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

Example command: `change hunt-group n`, where *n* is the assigned hunt group number.

Hunt group: page 1



ACD

Indicates whether or not to use Automatic Call Distribution (ACD) for this hunt group. Available only if ACD is enabled for the system.

Valid Entry	Usage
y	The hunt group functions as an ACD split/skill. Hunt groups used for voice messaging can function as ACD splits/skills.
n	The hunt group does not function as an ACD split/skill. This option should be used if this hunt group is on a remote server running Communication Manager and using voice messaging in a Distributed Communications System (DCS).

Related Links

ACD on page 807

(Calls Warning) Extension

Extension used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Calls Warning Port** is assigned a valid port, then the extension is removed and unassigned.

This field cannot be blank.

Available only if a queue has been enabled for this hunt group, and if a port number is not administered for the calls warning and time warning ports.

Related Links

(Calls Warning) Port on page 368
Queue on page 374
(Time Warning) Port on page 375

(Calls Warning) Port

The seven-character port number assigned to connect the optional external **Auxiliary Queue Call Warning Threshold** lamp that flashes when the number of calls in queue has exceeded the queue warning threshold (assigned in **Calls Warning Threshold**). Available only if a queue has been enabled for this hunt group.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number This port is assigned to an Analog Line circuit pack or given an x designation if an extension is used.

Example

01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Related Links

Calls Warning Threshold on page 368
Queue on page 374

Calls Warning Threshold

Available only if a queue has been enabled for this hunt group.

Valid Entry	Usage
1 to 999	The number of calls that can be queued before the system flashes the queue status (feature buttons assigned on agents telephones) and the optional **Auxiliary Queue Call Warning Threshold** lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and within the threshold. This value must be less than or equal to the queue length or left blank. If you administer the **Calls Warning Port** field, do not leave this field blank.

Related Links

Queue on page 374
(Calls Warning) Port on page 368

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number that reflects the required restriction for the hunt group. If this is a hunt group supporting voice messaging in a Distributed Communications System (DCS), the CORs on this screen for each server running Communication Manager must be the same.

Coverage Path

The coverage path for the hunt group. Available only if the hunt group is not vector controlled.

Valid Entry	Usage
digits	A coverage path number. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.
t1 to t999	Time of day table.

Related Links

Vector on page 376

Group Extension

An unused extension number assigned to the hunt group. This field cannot be blank.

Group Name

A character string that uniquely identifies the hunt group. Accepts up to 27 characters.

Note:

This field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW telephones.

For more information on Unicode language display, see “Administering Unicode Display” in *Administering Avaya Aura® Communication Manager* .

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Example

“parts dept”, “purchasing”, or “sales dept”

Related Links

Display Character Set on page 787

Group Number

The hunt group number.

Group Type

The group types of the hunt group depend on the system configuration. For more information about the availability of the group types, see the following table:

Hunt Group and Call Center options	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	sim
ACD = n, **Queue** = n, **Vector** = n	x	x						
ACD = y, **Vector** = n or y, **Skill** = n		x	x					
ACD = y, **Vector** = y, **Skill** = y			x	x				
ACD = y, **Vector** = y, **Skill** = y, **LOA** = y			x	x	x	x		
ACD = y, **Vector** = y, **Skill** = y, **Business Advocate** = y			x	x	x	x		

Table continues…

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Hunt Group and Call Center options	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	sim
ACD = y, **Vector** = y, **Skill** = y, **Dynamic Advocate** = y			x	x	x	x	x	
ACD = y, **Vector** = y, **Skill** = y, **Service Level Maximizer** = y								x

For a configuration that is not listed, the **LOA** field and the **Business Advocate** or the **Dynamic Advocate** customer options are disabled and therefore, the configuration does not apply.

Valid entry	Usage
circ	With the Circular call distribution, Communication Manager routes the calls to the administered extensions in a round-robin order. Communication Manager keeps track of the last extension in the hunt group to which a call was routed and routes the next call to the next extension in the circular list, independent of how long the extension has been idle. You cannot use this option with **ACD**, **Queues**, or **Vectors** enabled. The Circular option is available only with **ACD** disabled.
ddc	With the Direct Department Call (DDC) distribution, Communication Manager routes the calls to the first extension or ACD agent assigned in the ACD split. This option is also known as hot seat distribution. Communication Manager always routes an incoming call to the first available agent extension in the administered sequence. If the agent is unavailable, the call routes to the next available agent. Incoming calls are always routed to the first agent in the sequence, so calls are not evenly distributed among agents. You cannot use this option when the hunt group is administered as a skill. The DDC option is available only with the following configurations: • **ACD** disabled, or • **ACD** enabled, **Split** enabled or disabled, **Vector** enabled or disabled
ucd-mia	With the Uniform Call Distribution-Most Idle Agent (UCD-MIA) call distribution, Communication Manager routes calls to the most-idle agent, based on the agent

Table continues…

Valid entry	Usage
	occupancy, and adds the agent who finishes the most recent call to the end of the idle list. You can use this option if the hunt group is used for voice messaging. Enter the `ucd-mia` or the `ucd-loa` group types when the system supports the Outbound Call Management feature and **Controlling Adjunct type** is asai. The UCD-MIA option is available only with **ACD** enabled.
ucd-loa	With the Uniform Call Distribution-Least Occupied Agent (UCD-LOA) call distribution, Communication Manager routes calls to the least occupied agent, based on agent occupancy. You can use this option if the hunt group is used for voice messaging. Enter the `ucd-mia` or the `ucd-loa` group types when the system supports the **Outbound Call Management** feature and when the **Controlling Adjunct type** is `asai`. The UCD-LOA option is available only with **LOA** or **Business Advocate** enabled.
ead-mia	With the Expert Agent Distribution-Most Idle Agent (EAD-MIA) call distribution, Communication Manager routes calls to the agent with the highest skill level available for the call. If two or more agents with equal skill levels are available, Communication Manager routes the call to the most idle agent, based on when the agent finished the most recent call. The call is distributed to the agent best able to handle the call. The EAD-MIA option is available only with **EAS** and **Skill** enabled.
ead-loa	With the Expert Agent Distribution-Least Occupied Agent (EAD-LOA) call distribution, Communication Manager routes calls to the agent available with the highest skill level for the call. If two or more agents are available with equal skill levels, Communication Manager routes the call to the least occupied agent, based on agent occupancy. The call is distributed to an agent best able to handle the call. The EAD-LOA option is available only with **EAS**, **LOA**, or **Business Advocate** enabled.
pad	With the Percentage Allocation Distribution (PAD) call distribution, Communication Manager selects an agent from a group of available agents, based on a comparison of the work time, and the target allocation for the skill of an agent. The PAD option is available only with **Dynamic Advocate** enabled.
slm	With the Service Level Maximizer (SLM) call distribution, Communication Manager compares the current service level for each SLM-administered skill to a user-defined call service level target and identifies the skills that are most in need of agent resources to meet their target service level. Communication Managerthen identifies the available agents, assesses the overall opportunity cost of the agents, and selects only those agents whose other skills have the least need for their service at present. The SLM option is available only with **Service Level Maximizer** enabled and **Business Advocate** disabled.

Related Links

ACD on page 807
Business Advocate on page 808
Controlling Adjunct on page 377
Expert Agent Selection (EAS) on page 808

ISDN/SIP Caller Disp

This field is required for **ISDN-PRI, ISDN-BRI, and SIP Trunks**.

Valid Entry	Usage
grp-name	The hunt group name is sent to the originating user.
mbr-name	The member name is sent to the originating user.
blank	Displays the VDN name to the originating user.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Local Agent Preference

Enables or disables Local Agent Preference that routes an incoming ACD call to an idle agent by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. Available only with Call Center Release 3.0 or later when Expert Agent Selection (EAS) and multiple locations are enabled for the system. Also, the hunt group must be administered as a skill hunt group.

Related Links

Call Center Release on page 808
Expert Agent Selection (EAS) on page 808
Multiple Locations on page 804
Skill on page 383

MM Early Answer

Enables or disables MM Early Answer. The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue. This field applies only for systems using Multimedia Call Handling.

Related Links

Multimedia Call Handling (Basic) on page 803

Night Service Destination

Unavailable for vector-controlled hunt group.

Valid Entry	Usage
An assigned extension number	The destination where calls to this split redirect when the split is in the night service mode. This extension can be a VDN extension. Must be a local extension for all features to work correctly.
attd	An attendant group code.

Related Links

Vector on page 376

Queue

Enables or disables a queue for the hunt group.

Queue Limit

Available only if a queue is enabled for the hunt group.

Valid Entry	Usage
1 to 999	The limit to the number of calls that will queue.
unlimited	The system dynamically allocates the queue slots from a common pool on an as-needed basis.

Related Links

Queue on page 374

Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

Time Warning Extension

Extension used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Time Warning Port** is assigned a valid port, then the extension is removed and considered unassigned.

Available only if a queue is enabled for the hunt group. Required if a port number is not administered for the time warning port.

Related Links

Queue on page 374
(Time Warning) Port on page 375

(Time Warning) Port

The port number assigned to the **Auxiliary Queue Time Warning** lamp that flashes when the Time Warning Threshold has been reached by a call in queue. Available only if a queue has been enabled for the hunt group.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number This port is assigned to an Analog Line circuit pack or given an X designation if an extension is used.

Example

01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Related Links

Queue on page 374
Time Warning Threshold on page 375

Time Warning Threshold

Available only if a queue is enabled for the hunt group and if a port number is not administered for the call warning and time warning ports.

Valid Entry	Usage
0 to 999	The time in seconds that a call can remain in the queue before the system flashes the **Queue** status lamps (feature buttons assigned members telephones) and the **Auxiliary Queue Time Warning** lamp assigned to this split/skill. An entry of 0 provides a warning whenever a call is queued.

Related Links

Queue on page 374
(Time Warning) Port on page 375

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Vector

Enables or disables this hunt group as vector controlled. Available only if Basic Vectoring is enabled for the system.

Related Links

Vectoring (Basic) on page 810

Hunt group: page 2

This screen can vary according to values for particular fields on the previous page.

If the **ACD** is disabled for the system, this page is omitted.

```
change hunt-group 1                                          Page   2 of  62
                                  HUNT GROUP

                    Skill? n      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none
     Supervisor Extension:
   Priority On Intraflow? n

     Controlling Adjunct: none



  Multiple Call Handling: none

Timed ACW Interval (sec):      After Xfer or Held Call Drops? n
```


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AAS

Enables or disables this hunt group serving as an Auto-Available Split (AAS). With AAS, members of an ACD split or skill can be in auto-in work mode continuously. An agent in auto-in work mode becomes available for another ACD call immediately after disconnecting from an ACD call. Available only if ACD is enabled for this hunt group.

Related Links

ACD on page 367

Adjunct CTI Link

The ASAI CTI Link. This field cannot be blank. Available only if ACD is enabled for this hunt group, and the **Controlling Adjunct** is asai or adjlk.

Related Links

ACD on page 367
Controlling Adjunct on page 377

Controlling Adjunct

Available only if ACD is enabled for the hunt group. ASAI Link Core Capabilities and Computer Telephony Adjunct Links must be enabled for a value other than none.

Valid Entry	Usage
none	Members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can use only their data terminal keyboards to perform telephone functions (for example, change work state). Use if the controlling adjunct is a CONVERSANT voice system.
adjlk	Computer Telephony Adjunct Links
asai-ip	ASAI links administered without hardware.
adj-ip	ASAI adjunct links administered without hardware.

Related Links

ACD on page 367
ASAI Link Core Capabilities on page 795
Computer Telephony Adjunct Links on page 797

Dynamic Percentage Adjustment

Enables or disables automatic adjustments to agents' target allocations as needed to help meet the administered service level targets. Available only if ACD is enabled for the hunt group and this is a Percent Allocation Distribution (PAD) hunt group. Requires Business Advocate software.

Related Links

ACD on page 367
Business Advocate on page 808
Group Type on page 370

Dynamic Queue Position

Enables or disables dynamic queue operation to the calls queued to the skill. Dynamic Queue Position is a Business Advocate feature for calls to queue from multiple VDNs to a single skill, while maintaining different service objectives for those VDNs. Available only if ACD, Expert Agent Selection (EAS), and Skill are enabled for the hunt group. Requires Business Advocate software.

Related Links

ACD on page 367
Business Advocate on page 808
Skill on page 383

Dynamic Threshold Adjustment

Enables or disables automatic adjustments to overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels. Available only if ACD and Service Level Supervisor are enabled for the hunt group. Requires Business Advocate software.

Related Links

ACD on page 367
Business Advocate on page 808
Service Level Supervisor on page 382

Expected Call Handling Time (sec)

Available only if ACD is enabled for the system and if either Vectoring (Advanced Routing) or Business Advocate is enabled for the system.

Valid Entry	Usage
1 to 9999 in increments of 1	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the Business Advocate Percent Allocation feature.

Related Links

ACD on page 367
Business Advocate on page 808
Vectoring (G3V4 Advanced Routing) on page 811

Inflow Threshold (sec)

Available only if ACD and a queue are enabled for the system. unavailable for a vector-controlled hunt group.

Valid Entry	Usage
0 to 999	The number of seconds that a call can remain in the queue before no more calls are accepted by the queue. If 0 is entered, a call is redirected to this split/skill only if there is an available agent.

Related Links

ACD on page 367
Queue on page 374
Vector on page 376

Level 1 Threshold (sec)

Available only if ACD and Service Level Supervisor are enabled for the hunt group.

Valid Entry	Usage
0 to 60	The number of seconds for the first Expected Wait Time (EWT) threshold.

Example

If there are 45 calls whose EWT exceeds 45 seconds, threshold 1 will have been exceeded.

Related Links

Service Level Supervisor on page 382

Level 2 Threshold (sec)

Available only if ACD and Service Level Supervisor are enabled for the hunt group.

Valid Entry	Usage
0 to 60	The number of seconds for the second Expected Wait Time (EWT) threshold.

Example

If there are 60 calls whose EWT exceeds 60 seconds, threshold 2 will have been exceeded.

Related Links

ACD on page 367
Service Level Supervisor on page 382

Maximum Auto Reserve Agents

Available only if ACD is enabled for the hunt group and it is a Service Level Maximizer (SLM) type hunt group.

Valid Entry	Usage
0 to 9	The maximum number of Auto Reserve Agents available for this skill (hunt group). Any time an auto-reserve skill is in danger of falling below its target service level percent, some of this skill's agents are auto-reserved (kept idle in other skills) so that they are available when a new call arrives for this skill. Default is 0.

Related Links

Group Type on page 370

Measured

Provides measurement data for the ACD split/skill to VuStats or BCMS. Available only if ACD is enabled for the hunt group and VuStats or BCMS is enabled for the system.

Valid Entry	Usage
internal	Provides measurements made by the Call Management System that are internal to the server running Communication Manager.
external	Provides measurements made by the Call Management System that are external to the server running Communication Manager.
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Related Links

ACD on page 367
BCMS (Basic) on page 807
VuStats on page 812

Multiple Call Handling

Defines whether the hunt group can have multiple call handling capabilities, and if so, what type. Available only if ACD is enabled for the hunt group and Multiple Call Handling is enabled for the system.

Valid Entry	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the telephone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/ skill or direct agent call then is routed to the agent.
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/ Manual-In (MI/AI) work mode, and an unrestricted line appearance is available. Valid in an EAS environment for a skill-enabled hunt group.

Related Links

ACD on page 367
Multiple Call Handling (Forced) on page 809
Skill on page 383

Priority On Intraflow

Enables or disables having calls intraflowing from this split to a covering split and given priority over other calls waiting in the covering split queue. Available only if ACD field is enabled for the hunt group. Unavailable for a vector-controlled hunt group.

Related Links

ACD on page 367
Vector on page 376

Service Level Interval

The time interval when Actual Service Level (ASL) calculations run. ASL is one of the Service Level Maximizer (SLM) algorithms used for most situations, particularly for low staff or low traffic. The interval can be set to the same interval used when specifying the target objectives for the

application. Available only if Actual is administered for the SLM algorithm feature and this is an SLM-type hunt group.

Valid Entry	Usage
hourly	ASL algorithm calculations for accepted call and total call components are set to 0 at hourly intervals.
daily	ASL algorithm calculations for accepted call and total call components are set to 0 at daily intervals. This is the default.
weekly	ASL algorithm calculations for accepted call and total call components are set to 0 at weekly intervals. The weekly interval starts as 00:00 hours on Sunday.

Related Links

Service Level Maximizer Algorithm on page 320
Group Type on page 370

Service Level Supervisor

Enables or disables Service Level Supervisor for this skill. Service Level Supervisor is a Business Advocate feature that alleviates the need to move agents from skill to skill during emergencies or unanticipated peaks in call volume. Available only if ACD is enabled for the hunt group and this is an Expert Agent Selection (EAS) skill hunt group. Requires Business Advocate software.

Related Links

ACD on page 367
Skill on page 383
Business Advocate on page 808

Service Level Target (% in sec)

The system displays Service Level Target (% in sec) when the **ACD** field and the **Measured** field is not blank, and when one or more of the following features are active:

- **BCMS/VuStats Service Level** field on the System Parameters Customer-Options screen is active and the **Measured** field is set to internal or both. The service level target in seconds is used as the acceptable level for reporting the percentage of calls answered within the specified time. The percentage can be set to the default of 80%.
- Business Advocate on the System Parameters Customer-Options screen is active. The service level target in seconds is used for the Business Advocate Service Level Supervisor service level objective. This service level target can also be used for the dynamic percentage adjustment when the **Dynamic Threshold Adjustment** field on the Hunt Group screen is y and for the dynamic percent adjustment when the **Group Type** field on the Hunt Group screen is pad and the **Dynamic Percent Adjustment** field on the Hunt Group screen is set to y.
- The system displays the **Service Level Target** field when the **Group Type** field on the Hunt Group screen is slm, and on the System Parameters Customer-Options screen, the Service

Level Maximizer is active, and the Business Advocate customer option license is inactive. In this case the setting is also used as the service level target to trigger SLM.

- **Interruptible Aux Threshold** field on the Hunt Group screen is set to service-level-target. The Interrupt Aux feature is triggered if the service level drops below the administered percentage of calls in the specified seconds.

Valid Entry	Usage
1 to 99 (percentage)	The percentage component of the service level target. The default value is 80%.
1 to 9999 (time in seconds)	The time component of the service level target. The default value is 20 seconds.

Related Links

ACD on page 367
BCMS (Basic) on page 807
Business Advocate on page 808
Dynamic Percentage Adjustment on page 378
Dynamic Threshold Adjustment on page 378
Group Type on page 370
Interruptible Aux Threshold on page 387
Service Level Maximizer on page 810
VuStats on page 812

Service Objective

Available only if ACD is enabled for the hunt group and this is an Expert Agent Selection (EAS) skill hunt group. Requires Business Advocate software.

Valid Entry	Usage
1 to 9999	Sets a service objective for a specific skill as the number of seconds the call gets answered. The default value is 20.

Related Links

ACD on page 367
Business Advocate on page 808
Skill on page 383

Skill

Enables or disables this hunt group as an Expert Agent Selection (EAS) skill. Available only if ACD is enabled for the hunt group and EAS is enabled for the system.

Related Links

ACD on page 367
Group Type on page 370
Expert Agent Selection (EAS) on page 808

SLM Count Abandoned Calls

Available only if Actual is the administered Service Level Maximizer (SLM) algorithm for the feature and this is an SLM-type hunt group.

Valid Entry	Usage
y	Abandoned calls are included in the Actual Service Level (ASL) algorithm calculations for SLM.
n	Abandoned calls are excluded in the ASL algorithm calculations for SLM. This option is best used when reporting for this application does not account for calls that are abandoned while in skill queues.

Related Links

Group Type on page 370
Service Level Algorithm for SLM on page 325

Supervisor Extension

The extension number of the ACD split/ skill supervisor that agents reach when using the Supervisor Assist feature. The extension number cannot be a Vector Directory Number (VDN).

Available only if ACD field is enabled for the system.

Related Links

ACD on page 367

Timed ACW Interval (sec)

Available only if ACD is enabled for the hunt group and Timed ACW is enabled for the system.

Note:

This field can be overridden by the settings administered for a vector. Coordinate the settings when setting up delays.

Valid Entry	Usage
1 to 9999	The number of seconds an agent in auto-in work mode remains in After Call Work (ACW) mode after a call drops. After this time interval expires, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split.

Related Links

ACD on page 367
Timed ACW on page 810

VuStats Objective

Available only if ACD is enabled for the hunt group and VuStats is enabled for the system. Also, the hunt group must be administered to collect internal or both internal and external measurement data for VuStats.

Valid Entry	Usage
0 to 99999	A numerical objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The VuStats screen displays the objective. The agents and supervisors can compare the current performance against the value of the objective for the split or skill. This value applies to customized VuStats display formats.

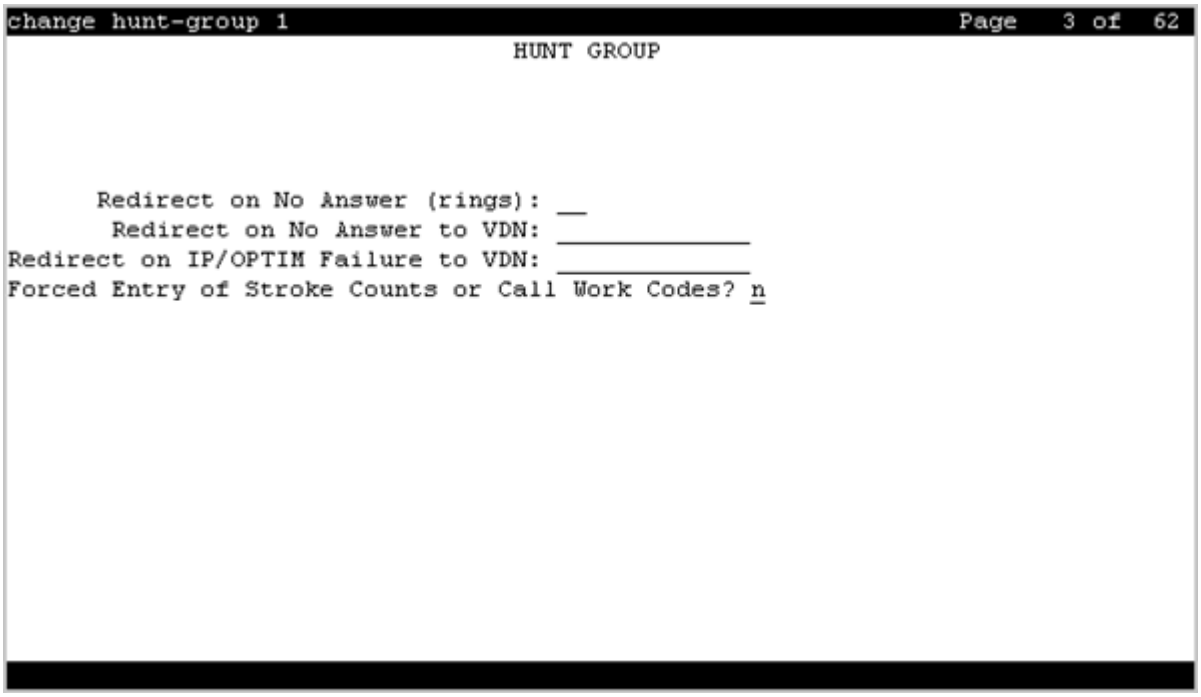
Related Links

ACD on page 367
Measured on page 380
VuStats on page 812

Hunt group: page 3

This screen can vary according to values for particular fields on the previous page.

If the **ACD** is disabled for the system, this page is omitted.



Forced Entry of Stroke Counts or Call Work Codes

Enables or disables the requirement that either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode. Available only if ACD is enabled for the hunt group and if the hunt group does *not* have a Controlling Adjunct.

Related Links

ACD on page 367
Controlling Adjunct on page 377

Interruptible Aux Deactivation Threshold

This field specifies the maximum level of service above which the system deactivates the Interruptible Aux feature.

Use the **Interruptible Aux threshold** and the **Interruptible Aux Deactivation Threshold** fields to maintain a buffer between the two levels of service that trigger the **Interruptible Aux** feature. The values of the **Interruptible Aux threshold** and the **Interruptible Aux Deactivation Threshold** fields must have a difference of one unit.

The valid entries for this field depend on the value of the **Interruptible Aux threshold** field:

- If you set the value of the **Interruptible Aux threshold** field to calls-warning-threshold, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 998.

- If you set the value of the **Interruptible Aux threshold** field to service-level-target, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 100.
- If you set the value of the **Interruptible Aux threshold** field to time-warning-threshold, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 998.
- If you set the value of the **Interruptible Aux threshold** field to none, the system does not display the **Interruptible Aux Deactivation Threshold** field.

Interruptible Aux Threshold

This field specifies the minimum level of service below which the system enables the Interruptible Aux feature. Use the Interruptible Aux feature to make the Expert Agent Selection (EAS) agents in the AUX work mode available to receive calls. This feature works only if the agents have an interruptible reason code.

Valid entry	Usage
service-level-target	Use this option to enable the Interruptible Aux feature when the service level drops below the administered percentage of calls within the specified period. For example, if you set the **Service Level Target (% in sec)** field to `90`% of calls in `30` seconds, the system enables the Interruptible Aux feature if the service drops to 89% of calls in 30 seconds.
calls-warning-threshold	Use this option to enable the Interruptible Aux feature when the number of calls in the queue for a hunt group exceeds the specified number of calls. For example, if you set the **Calls Warning Threshold** field to `20`, the system enables the Interruptible Aux feature if the number of calls in the queue exceeds 20.
time-warning-threshold	Use this option to enable the Interruptible Aux feature when the oldest call is in the queue for longer than the specified number of seconds. For example, if you set the **Time Warning Threshold** field to `60` seconds, the system enables the Interruptible Aux feature if the oldest call is in the queue for more than 60 seconds.
none	Use this option if you do not want to administer the Interruptible Aux feature.

Redirect on IP/OPTIM Failure to VDN

A blank in the field redirects the call back to the hunt group. VDN extension redirects to the specified VDN.

If **Redirect on IP/OPTIM Failure to VDN** is not assigned, the call is re-queued to the same skill at a high priority. If there are no queue slots available, the caller will hear a busy signal. If all fails, the caller receives ring back until the system receives a caller disconnect.

Related Links

ACD on page 367

Redirect on No Answer (rings)

Available only if ACD is enabled for the hunt group.

Valid Entry	Usage
1 to 20	The maximum number of rings before a call redirects back to the split/skill, or to the administered VDN.
blank	Deactivates Redirect on No Answer.

Related Links

ACD on page 367

Redirect on No Answer to VDN

The extension number of the VDN used to redirect a Redirect On No Answer (RONA) call to a VDN instead of to the split/skill. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that routes to an off-premises VDN.

Direct Agent calls go to the agent's coverage path if it is administered. If not, the calls go to a VDN.

Available only if ACD is enabled for the hunt group. Requires administration of the number of rings before a call will redirect.

Related Links

ACD on page 367
Redirect on No Answer (rings) on page 388

Hunt Group: page 4

The Hunt Group Message Center screen can vary according to system configuration and values populating particular fields.

```
change hunt-group 1                                          Page   4 of  62
                                  HUNT GROUP

                      LWC Reception: none            AUDIX Name: _______
          First Announcement Extension: _______

                      Message Center: none_______
```

AUDIX Name

The name of the AUDIX machine. Must be the same name as the IP Node name and administered *after* the IP Node is configured.

Related Links

IP Node Names on page 451

IP Node Names on page 451

Calling Party Number to INTUITY AUDIX

The system displays Calling Party Number to INTUITY AUDIX when the **Message Center** field is audix or rem-vm.

Valid entries	Usage
y/n	Enter y to send the calling party's number to INTUITY AUDIX®

Related Links

Message Center on page 391

First Announcement Delay (sec)

Available only if a queue is administered for the hunt group. Unavailable if the hunt group is vector controlled.

Valid Entry	Usage
0 to 99	The number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call has not been answered after the announcement, the caller hears music for first announcement only if Music-on-Hold is provided, or ringing for as long as the call remains in queue. When 0 is entered, the first announcement is provided immediately to the caller. This value is set automatically to 0 if there is no queue.
blank	There is no first announcement.

Related Links

Queue on page 374
Vector on page 376

First Announcement Extension

The recorded announcement extension number the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music only after the first announcement if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement. Otherwise, the caller hears music, if provided.

Available only if ACD and a queue is administered for the hunt group. Unavailable if the hunt group is vector controlled.

Note:

If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Related Links

ACD on page 367

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related Links

AUDIX Name on page 389

Message Center

The type of messaging adjunct for the hunt group. For information on system capacities, see the *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Valid Entry	Usage
audix	AUDIX located on this server running Communication Manager
fp-mwi	Public network allowing AUDIX to be located on another switch. Available only if ISDN Feature Plus is administered for the system.
msa	Messaging Server Adjunct
msa-vm	A voice mail system integrated using Mode Codes or Digital Station Emulation
rem-vm	DCS feature allowing voice mail to be located on another server
qsig-mwi	QSIG network allowing voice mail to be located on another server
sip-adjunct	SIP message center server
none	The hunt group does not serve as a message hunt group.

Related Links

ISDN Feature Plus on page 801

Message Center AUDIX Name

The name of the Message Center AUDIX. Available only if the messaging type is audix or rem-vm.

Related Links

Message Center on page 391

Message Center MSA Name

The name of the Message Center Messaging Server Adjunct (MSA). Available only if the messaging type is msa.

Related Links

Message Center on page 391

Message Center AUDIX Name on page 391

Primary

Available only if the **Message Center** field is audix or rem-audix.

Valid entries	Usage
y/n	Enter y to indicate that the specified messaging server is the primary adjunct.

Related Links

Message Center on page 391

Provide Ringback

Enables or disables ringback to the calling party until a **Connect** is received for the call to the Messaging system. Ringback is discontinued upon receipt of the **Connect** indication. Used for an SBS trunk for the QSIG MWI hunt group. A call covering to the message center provides ringback to the caller during the coverage interval. Available only if the messaging type is fp-mwi or qsig-mwi.

Related Links

Message Center on page 391

Routing Digits (e.g. AAR/ARS Access Code)

A one- to four-digit AAR (qsig-mwi) or ARS (fp-mwi) access code. This access code is prepended to the AUDIX Complete Number to define a route to the Message Center switch hunt group containing the line ports to the AUDIX. Accepts characters * and #.

Available only if the messaging type is qsig-mwi or fp-mwi.

Related Links

Message Center on page 391

Second Announcement Delay (sec)

Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Valid Entry	Usage
1 to 99	The time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated. If this split/skill or hunt

Table continues…

Valid Entry	Usage
	group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	There is no second announcement.

Related Links

ACD on page 367
Queue on page 374
Vector on page 376

Second Announcement Extension

The extension number assigned to a second recorded announcement. Left blank if there is no second announcement.

Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Note:

If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Related Links

ACD on page 367
Queue on page 374
Vector on page 376

Second Announcement Recurring

Allows or disallows repeating the second announcement. Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Related Links

ACD on page 367
Queue on page 374
Vector on page 376

Send Reroute Request

Allows or disallows rerouting getting invoked when a call covers through a qsig-mwi hunt group. Available only if the messaging type is qsig-mwi and Supplementary Services with Rerouting is enabled for the system.

Related Links

Message Center on page 391
Supplementary Services with Rerouting on page 814

TSC per MWI Interrogation

Controls Temporary Signaling Connections (TSCs) used for message waiting interrogations for users that are “local” to the system in which the hunt group is administered. Available only if the messaging type is qsig-mwi.

Valid Entry	Usage
y	Communication Manager brings the TSC up, executes the Interrogate operation, and then tears the TSC down.
n	Communication Manager uses the existing TSC sending FACILITY messages to request MWI status if the TSC is already set up, or sets up a TSC. When the interrogation operation is complete, leaves the TSC up, subject to the existing timer. This is the default.

Related Links

Message Center on page 391

Voice Mail Extension

The UDP extension of the voice mail hunt group on the host server running Communication Manager.

Available only if the messaging type is rem-vm.

Related Links

Message Center on page 391

Voice Mail Handle

The Session Manager (SM) handle that can receive voice mail. Can be left blank if a Voice Mail Number has been assigned.

Voice Mail Number

The 1- to 17-digit voice mail dial-up number. The qsig-mwi selection shows the complete number of the AUDIX hunt group on the Message Center server for QSIG MWI. The fp-mwi selection shows the public network number of the AUDIX hunt group on the Message Center server.

Available only if Basic Call Setup and Basic Supplementary Services are enabled for the system, and the messaging type is qsig-mwi or fp-mwi.

Related Links

Basic Call Setup on page 813
Basic Supplementary Services on page 814
Message Center on page 391

Hunt Group: page 5 through X

```
change hunt-group 1                                          Page   5 of  62
                                HUNT GROUP
         Group Number: 1    Group Extension: 369-9999      Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /1
                                           Total Administered Members: 1
GROUP MEMBER ASSIGNMENTS
      Ext             Name(19 characters)        Ext             Name(19 characters)
   1: 360-5003        Kaliya 3605003          14:
   2:                                         15:
   3:                                         16:
   4:                                         17:
   5:                                         18:
   6:                                         19:
   7:                                         20:
   8:                                         21:
   9:                                         22:
  10:                                         23:
  11:                                         24:
  12:                                         25:
  13:                                         26:

  At End of Member List
```

Administered Members (min/max)

The minimum and maximum member number administered for this hunt group. Available for all member pages.

At End of Member List

Displays the current page as also the last page.

Group Extension

The extension of the hunt group.

Group Number

Displays the number of the hunt group.

Group Type

Displays the type of hunt group.

Member Range Allowed

The range of allowed members. These values vary depending on the system or configuration.

More Members Exist

Statement that there are more members and more pages than currently displayed.

Total Administered Members

The total number of members administered for the hunt group.

GROUP MEMBER ASSIGNMENTS

Ext

The assigned station or attendant console extension. This extension cannot be a Vector Directory Number (VDN). The data module cannot be a member of an ACD split/skill. Administers the assigned station or attendant console extension only if the controlling adjunct is administered as

none. Displays the assigned station or attendant console extension if the controlling adjunct is administered as asai.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Controlling Adjunct on page 377

Name

The name assigned to the extension number when it is administered in the system.

Related Links

Ext on page 396

Chapter 59: Incoming Call Handling Treatment

Incoming Call Handling Treatment (ICHT) specifies a unique call treatment for incoming calls on ISDN and SIP trunk groups. This screen is available only if digit handling is administered as overlap, and the direction is set to outgoing.

Note:

ICHT is also used to modify the Calling Party number when Communication Manager is used as a feature server.

The Incoming Call Handling Treatment Table screen displays entries with a predefined service or feature before entries with a user-defined service or feature .

The user-defined entries are always listed in the reverse order on Incoming Call Handling Treatment Table as compared to the entries that show on Network Facilities. Therefore, for two user-defined services or features, ABC and XYZ, you can force XYZ to show before ABC on Incoming Call Handling Treatment Table by administering XYZ after ABC on Network Facilities.

Note:

DCS features that use the **remote-tgs** button on the remote server do not work when the local trunk group deletes or inserts digits in the incoming call.

To change the value of the fields on the Incoming Call Handling Treatment Table, use the following command:

`change inc-call-handling-trmt trunk-group` *n*, where *n* is an ISDN or a SIP trunk group.

Called Len

Valid Entry	Usage
0 to 21	The number of digits received for an incoming call. Zero is used when the Public Switched Telephone Network (PSTN) provider does not provide any “Number Digits” within the received **Called Party IE**, such as in Japan.
blank	When **Called Number** has also been set to blank, so that any length of digits associated with the **Called Party IE** of the Incoming SETUP message matches this field.

Called Number

Valid Entry	Usage
1 to 16	The number of leading digits received for an incoming call.
blank	Used as a “wildcard”, so that any number associated with the specified service or feature can match in this field.

Del

Valid Entry	Usage
1 to 21 blank	The number of leading digits to be deleted from the incoming **Called Party Number**. Calls of a particular type can be administered to be routed to a single destination by deleting all incoming digits and then administering the **Insert** field with the required extension.

Insert

Valid Entry	Usage
1 to 16 * #	The number of digits prepended to the front of the remaining digits after any optional digit deletions have been performed. The resultant number formed from digit deletion and insertion is used to route the call, provided night service is not in effect.

Per Call CPN/BN

Specifies when and how to request **Calling Party Number** (CPN) or **Billing Number** (BN) for calls of this type. Available only with ISDN trunk groups.

Valid Entry	Usage
cpn-only	Calling party number only
bn-only	Billing number only
bn-pref	Prefers billing number, but accepts calling party number
cpn-pref	Prefers calling party number, but accepts billing number
none	Communication Manager will not request either CPN or BN for any incoming calls of this type.

Table continues…

Valid Entry	Usage
blank	Leave blank when connected to another media server or switch, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that prohibits customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles “CPN/BN to Terminating End on a Per-Call Basis” and “ANI (BN) on Request”.

Note:

A 4-second delay occurs in terminating the call to the far-end station if the connecting server or switch stops responding to the request.

Night Serv

Available only for ISDN trunk groups.

Valid Entry	Usage
Assigned extension *Attendant group access code* attd blank	Specifies a night service extension per Service/Feature. A Vector Directory Number (VDN) is acceptable. This entry is overridden by the night service administered for an individual trunk or the trunk group.

Related Links

Night Service on page 870
Night on page 520

Service/Feature

Displays the number that corresponds to the server type administered for the trunk group or network facility. Administration is required for a call-by-call service type.

Valid Entry	Usage
Valid pre-defined Service/Feature values for cbc trunk groups	These values are administered for Network Facilities.

Table continues…

Valid Entry	Usage
User-defined facility type administered for the service type	• 0 = feature • 1 = service • 2 = incoming • 3 = outgoing

Related Links

Network Facilities on page 600
Facility Type on page 600

Chapter 60: Integrated Announcement Boards

Moves integrated announcement boards that have been previously administered on the Announcements/Audio Sources screen to a new board location. Displays a list of all administered integrated announcement circuit packs.

Example command: `display integrated-annc-boards`

```
display integrated-annc-boards                                  Page   1 of  13
                          INTEGRATED ANNOUNCEMENT BOARDS

     Brd        Tim       Num  Chk              Brd        Tim       Num  Chk
     Loc    Sfx Rem    Rt Annc Sum              Loc    Sfx Rem    Rt Annc Sum

  1:                                      16:
  2:                                      17:
  3:                                      18:
  4:                                      19:
  5:                                      20:
  6:                                      21:
  7:                                      22:
  8:                                      23:
  9:                                      24:
 10:                                      25:
 11:                                      26:
 12:                                      27:
 13:                                      28:
 14:                                      29:
 15:                                      30:
```

Board Location

The physical location of the integrated announcement circuit pack (UUCSS).

Checksum ID

Applies to TN750 only; not applicable to VAL.

Last Board Location Saved

Displays last board location saved. Applies to TN750 only; not applicable to VAL.

Number of Recordings

The number of non-zero-length announcement recordings or files on the circuit pack.

Rate

The announcement’s compression rate.

Sfx

The circuit pack suffix letters.

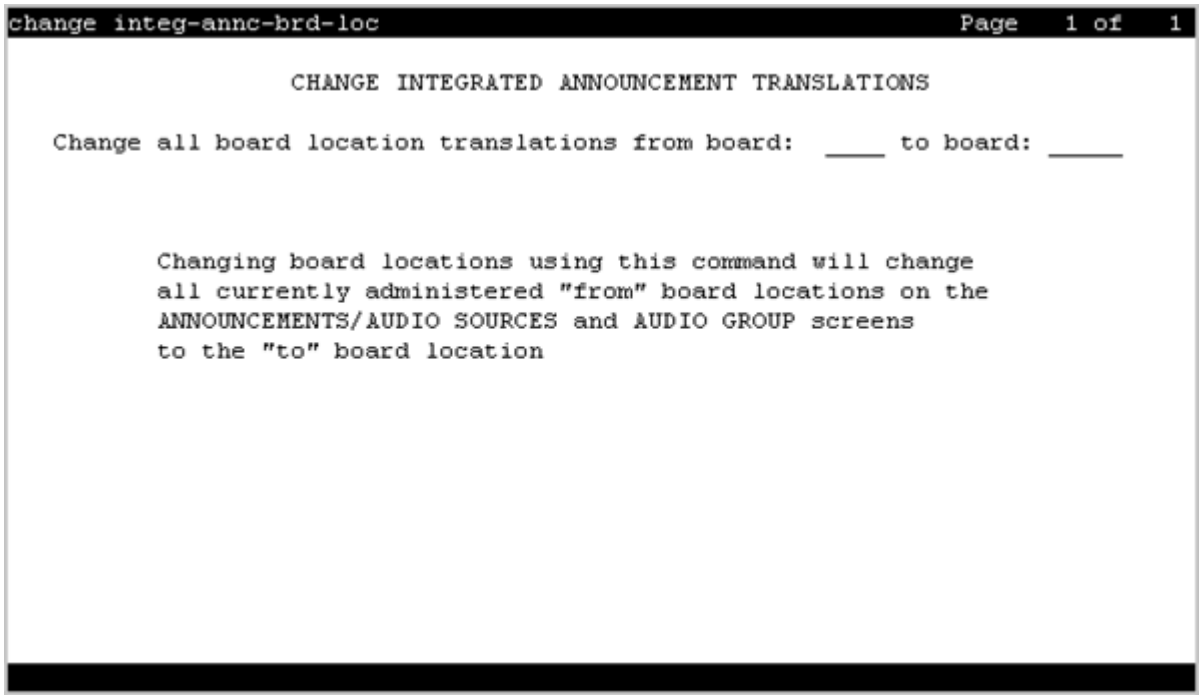
Time Remaining

The amount of recording time in seconds remaining on the circuit pack at the 64Kb rate.

Chapter 61: Integrated Announcement Translations

Changes board locations currently administered on the Announcements/ Audio Sources screen to a new board location.

Example command: `change integ-annc-brd-loc`



Change all board location translations from board

Valid Entry	Usage
board cabinet 1 to 3 carrier A to E	The VAL board that is currently administered.

Valid Entry	Usage
slot 1 to 20 or gateway 1 to 10 module V1 to V9	

to board

Use this field to specify the VAL board for announcement translations.

You can use one of the following ways to enter a value in this field:

- board number
- cabinet 1 to 3: carrier A to E: slot 1 to 20
- gateway 1 to 10: module V1 to V9

Chapter 62: Intercom Group

Assigns extensions to intercom groups.

Example command: `change intercom-group n`, where *n* is the assigned intercom group number.

DC

Assigns a dial code to an extension. Accepts up to two digits. The dial code is the code users must dial to make intercom calls to the corresponding extension. The number of digits entered must exactly match the number administered for the length of the dial code. This field cannot be blank.

Example

If the length of the dial code is set to 2, type `1` as 01 in the **DC** field.

Related Links

Length of Dial Code on page 407

Ext

Assigns an extension to the group. A Vector Directory Number (VDN) cannot be used as an extension.

Group Number

The group ID number.

Length of Dial Code

Sets the number of digits that users must dial to access an extension in the group.

Valid Entry	Usage
1	For nine or fewer members.
2	For 10 or more members.

Name

The name associated with the extension that has been administered to join the group.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Name on page 728

Chapter 63: Inter-Exchange Carrier (IXC) Codes

Identifies the IXC in the Call Detail Recording (CDR).

Example command: `change ixc-codes`

Inter-Exchange Carrier (IXC) Codes: page 1

IXC Access Number

The digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. You cannot use duplicate access numbers in the table. Accepts from 2 to 11 digits and the * character.

IXC Name

A description that identifies the IXC. Accepts up to 15 characters.

Inter-Exchange Carrier (IXC) Codes: page 2

IXC Code Format

A one- to four-digit IXC code format. Includes *, x, X, xxxx (for line 1), and xxx (for line 2).

IXC Prefix

A one- to three-digit prefix. Includes *, 101 (for line 1) and 10 (for line 2).

Chapter 64: Intra-Switch CDR

Administers extensions for which Intra-Switch Call Detail Recording (CDR) is enabled.

Note:

Attendants are disallowed to be optioned for the Intra-Switch CDR feature.

Example command: `change intra-switch-cdr n`, where *n* is the assigned extension number.

Assigned Members

The number of extensions currently administered for Intra-switch CDR.

Extension

The local extensions used to track with Intra-Switch CDR. The number of tracked extensions can vary by system.

Chapter 65: IP Address Mapping

Defines feature characteristics that depend on the IP address.

Example command: `change ip-network-map`

Emergency Location Extension

The emergency location extension for this station. Accepts up to thirteen digits. Allows the system to properly identify the location of a caller who dials a 911 emergency call from this station. An entry in this field must be of an extension type included in the dial plan, but does not have to be an extension on the local system. It can be a Uniform Dial Plan (UDP) extension. A blank entry is typically used for an IP softphone dialing in through PPP from somewhere outside the network.

For administered emergency numbers, the feature functions as follows:

- If the emergency location extension administered on the station screen is the same as the emergency location extension administered on the IP Address Mapping screen, the feature sends the station's own extension to the Public Safety Answering Point (PSAP).
- If the emergency location extension for the station is different from the emergency location extension administered here, the feature sends the extension administered here to the PSAP.

Caution:

On the ARS Digit Analysis Table, administer 911 to be call type emer or alrt in order for the E911 Emergency feature to work properly.

Related Links

Emergency Location Ext on page 716

From IP Address

The starting IP address. IPv6 format is supported.

Network Region

Valid Entry	Usage
1 to 2000	The network region for the IP address range. For SIP, the value for this field must correlate with the configured network region for this range of addresses. This field must contain a non-blank value if the corresponding **From IP Address** contains a non-blank value.

Subnet Bits

Valid Entry	Usage
0 to 64 blank	The number of bits of the subnet mask. Used in conjunction with the **From IP Address** to specify the end of the ID address range if a **To IP Address** is not administered.

To IP Address

The terminating IP address. IPv6 format is supported.

VLAN

Sends VLAN instructions to IP endpoints such as IP telephones and softphones. This field does not send VLAN instructions to the PROCR (S8300 Server), CLAN, and Media Processor boards.

Valid Entry	Usage
0 to 4094	The virtual LAN value.
n	Disabled

Chapter 66: IP codec set

Use this screen to specify the type of codec used for voice encoding and companding (compression/decompression).

The default codec is set for G711. The G711 provides the highest voice quality, but G711 uses the maximum bandwidth. If G711 does not meet the required voice-quality or bandwidth trade-off specification, you can change the G711 default setting.

To modify the values of the fields on this screen, use the following command:

`change ip-codec-set` *n*, where *n* is the IP codec set number.

IP codec set: page 1

Defines the allowed codecs and packet sizes used between VoIP resources. Enables silence suppression on a per-codec basis and dynamically displays the packet size in milliseconds for each codec in the set, based on the number of frames administered per packet.

Audio Codec

Specifies the audio codec used for this codec set.

- G.711A (a-law)
- G.711MU (mu-law)
- G.722-64k
- G.722.1-24k
- G.722.1-32k
- G.722.2
- G.723-5.3
- G.723-6.3
- G.726A-32K
- G.729
- G.729A

- G.729B
- G.729AB
- SIREN14-24k
- SIREN14-32k
- SIREN14-48k
- SIREN14-S48k
- SIREN14-S56k
- SIREN14-S64k
- SIREN14-S96k

Important:

Include at least two codecs for every telephone to avoid incompatible codecs. Use the codecs specified in the following table for the telephones shown.

Telephone	Codec to use
All Avaya IP Telephones	G.711, G.729B
4601 4602 4602SW 4620SW 4621SW 4622SW	add G.726A (requires firmware R2.2)

Codec Set

The number assigned to this Codec Set.

Frames Per Pkt

Specifies the number of frames per packet up to a packet size of 60 milliseconds (ms).

Valid Entry	Usage
1 to 6	Default frame sizes for codecs: • G.711 and G.729: 2 frames (20 ms) • G.723: 3 frames (30 ms) • G.726A: 1 frame (10 ms)

Media Encryption

Specifies a priority listing of the three possible options for the negotiation of encryption.Communication Manager attempts to provide bearer encryption per this administered priority order. The selected option for an IP codec set applies to all codecs defined in that set. Available only if Media Encryption over IP is enabled for the system.

Valid Entry	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: • Server-to-gateway (H.248) • Gateway-to-endpoint (H.323)
aea	Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: • All endpoints within a network region using this codec set must be encrypted. • All endpoints communicating between two network regions and administered to use this codec set must be encrypted. SRTP is a media encryption standard defined in RFC 3711 as a profile of RTP. Communication Manager supports the following functionality as given in RFC 3711: • Encryption of RTP (optional but recommended) • Authentication of RTCP streams (mandatory) • Authentication of RTP streams (optional but recommended) • Protection against replay
1-srtp-aescm128-hmac80	1-Encrypted/Authenticated RTP with 80-bit authentication tag
2-srtp-aescm128-hmac32	2-Encrypted/Authenticated RTP with 32-bit authentication tag
3-srtp-aescm128-hmac80-unauth	3-Encrypted RTP but not authenticated
4-srtp-aescm128-hmac32-unauth	4-Encrypted RTP but not authenticated
5-srtp-aescm128-hmac80-unenc	5-Authenticated RTP with 80-bit authentication tag but not encrypted
6-srtp-aescm128-hmac32-unenc	6-Authenticated RTP with 32-bit authentication tag but not encrypted
7-srtp-aescm128-hmac80-unenc-unauth	7-Unencrypted/Unauthenticated RTP

Table continues…

Comments on this document? infodev@avaya.com

Valid Entry	Usage
8-srtp-aescm128-hmac32-unenc-unauth	8-Unencrypted/Unauthenticated RTP **Note:** For stations, the only value supported is srtp-aescm128-hmac80. H.323 IP trunks support all eight of the listed algorithms.
none	Media stream is unencrypted. This is the default.

Packet Size (ms)

The packet size in milliseconds.

Silence Suppression

Enables or disables RTP-level silence suppression on the audio stream.

IP codec set: page 2

You can configure the following capabilities on the codec set:

- Enable or disable Direct-IP Multimedia for videophone transmissions.
- Enable or disable endpoints in the assigned network region to send fax, modem, or TTY calls over IP trunks.
- Define the mode that the system uses to route the fax, modem, or TTY calls.
- Determine whether redundancy must be added to the transmission for higher reliability and quality.

You must assign these capabilities to the codec set, and assign the codec set to a network region for endpoints in that region to be able to use the capabilities established on this screen.

Using V.150.1, you can configure fax and modem so that both use a relay mode together in an IP codec set. You can set the **FAX Mode** field to t.38–standard for fax calls to use t.38. However, modem calls use V.150.1.

Note:

Transporting modem signals over IP between Avaya Communication Manager systems and media gateways is a proprietary implementation. The selection of the V.150.1 Modem Relay codec choice is not proprietary. For modem transport interoperability with third-party gateways, v150mr is the recommended mode of operation.

Caution:

If users are using Super G3 fax machines and modems, do not assign these fax machines to a network region with an IP codec set that is modem-enabled as well as fax-enabled. If its **Codec set** is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

Allow Direct-IP Multimedia

To enable or disable direct multimedia, use the following codecs:

- H.261
- H.263
- H.264 (video)
- H.224
- H.224.1 (data, far-end camera control).

Clear-channel Mode

Enables or disables supporting this codec set for BRI data calls.

ECM

With **Error Correction Mode (ECM)**, fax endpoints correct the fax transmission error without the need to retransmit multiple pages.

The **ECM** field is available only when **FAX Mode** is t.38-standard or t.38-G711-fallback.

Valid entry	Usage
y	ECM-capable gateways use the T.38 protocol to relay T.30 ECM signaling and ECM frames between a local Group 3 fax machine and a far-end T.38 receiver. The default value is y.
n	ECM-capable gateways do not use the T.38 protocol to relay T.30 ECM signaling and ECM frames.

FAX Mode

Valid entry	Usage
off	For users using this mode, Communication Manager treats a fax call as an ordinary voice call.
relay	For users using this mode, Communication Manager uses the Avaya proprietary relay mode for fax transmissions over IP network facilities only between Avaya devices. This mode can be coupled with Redundancy.
pass-through	For users using this mode, Communication Manager uses the Avaya proprietary pass-through mode for in-band fax transmissions over IP network facilities only between Avaya devices. This mode can be coupled with Redundancy.
T.38-standard	For users using this mode, Communication Manager uses the T.38 protocol for fax transmission over IP network facilities. Communication Manager supports the transition of an existing audio call to a fax call, for both H.323 and SIP calls.
T.38-G711-fallback	For users using this mode, Communication Manager uses the T.38 protocol for fax transmission only if the protocol can be successfully negotiated with the peer SIP entity. Otherwise, Communication Manager falls back to G.711 for fax transmission. This mode requires a G.711 codec to be administered on the ip-codec-set screen.

Note:

If you have a telephone on an IP trunk that is close to a fax machine, the telephone can change into the fax mode by picking up the tones from the fax machine . To prevent the telephone from changing into the fax mode, set the **FAX mode** to off and put the fax machines in an Automatic Route Selection (ARS) partition that uses only circuit switched trunks, even for intergateway fax calls.

Maximum Call Rate for Priority Direct-IP Multimedia

The system displays this field only if **Allow Direct-IP Multimedia** is set to y.

Maximum Call Rate for Direct-IP Multimedia

Use this field to limit the size of bandwidth used for video calls. The system displays this field appears only if **Allow Direct-IP Multimedia** is set to y. The default value is 384 kilobits. You can enter a value between 64 kilobits and 15360 kilobits, with an increment of 64 kilobits. A typical video endpoint does not require more than 2048 kilobits.

Modem Mode

Valid entry	Usage
off	In the off mode, a modem call is treated as an ordinary voice call. The default mode for new installations and upgrades is off.
relay	In the Avaya proprietary relay mode, Avaya devices detect the tones of the call and use the appropriate modulation protocol or TTY to terminate or originate the modem calls over IP network.
pass-through	In the Avaya proprietary pass-through mode, Avaya devices detect the tones of the modem call and use G.711 encoding to process the call over the IP network.
v150mr	In the v150mr mode, Avaya devices detect the tones of the modem call and use V. 150.1 protocol to transmit V-series modem signals between secure terminals and third-party SIP gateways. **Note:** V.150.1 protocol is a standard recommended by International Telecommunication Union (ITU) to use a modem over IP networks that support dialup modem calls.

Packet Size

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The **Packet Size** column supports choices of 10, 20, 30, 40, 50, or 60 millisecond IP packet size to be used in the RFC 4040 transport.

The default choice is `20` milliseconds.

If the packet size is of value other than 20 milliseconds, then the SDP Offer must include an Attribute format string.

For example, for an administered Packet Size of 10 milliseconds, Communication Manager sends an SDP Offer message that includes the attribute string of a=ptime 10.

Redundancy

Use this field to assign the number of duplicate or redundant packets that must be sent in addition to the primary packet. You can enter a value from 1 to 3. The default value is 0.

Note:

For the **pass-through** and the **clear-channel** modes, you can use the default value or assign 1. This field does not apply to the v150mr mode. If you set the **Modem Mode** field to v150mr, the system sets the **Redundancy** field to display-only with the default value.

The **Redundancy** field supports choices of 0 and 1 for the **SIP64K** field. 0 designates that no redundant media packets are transmitted. 1 designates that a single redundant packet should be transmitted for each media stream packet that is sent.

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

SIP 64K Data

With SIP 64K Data, Communication Manager controls the mechanism to enable the support of the RFC 4040 media service.

Valid Entry	Usage
y	Communication Manager searches for an RFC 4040 capable gateway, G450 or G430. If the given gateway does not support the RFC 4040 feature, Communication Manager searches for an available gateway in the connected network region. If Communication Manager cannot detect a suitable gateway within the network region, the call request is dropped.
n	The default value is n.

Note:

The G650 gateway does not support the RFC 4040 feature. G430 and G450 gateways support the RFC 4040 feature. The gateway in the network region must support the RFC 4040 feature for Communication Manager to connect RFC 4040 calls.

Communication Manager uses RFC 4040 Clear Mode data transport to support the media transport of ISDN traffic. The ISDN traffic is directed to a destination that is reached through a SIP trunk.

Associated with the **SIP 64K Data** field are two other fields, **Redundancy** and **Packet Size (ms)**. Communication Manager communicates the values of the **Redundancy** and **Packet Size (ms)** fields to the media gateway so that the gateway properly operates with the DSP conversion of the TDM media into an IP media stream.

TDD/TTY Mode

Valid entry	Usage
off	Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting must send TTY calls to non-Avaya systems. However, there might be errors in character transmissions.
US	For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default value for new installations and upgrades.
UK	For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities.
pass-through	For users in regions using this codec, use the Avaya proprietary pass-through mode for TTY transmissions over IP network facilities.

IP codec set: page 3

The system displays this page only when you set the **Modem Mode** field to v150mr.

Modem Relay Preferred (NoAudio)

If you set this field to y, Communication Manager includes the NoAudio codec in the SIP SDP offer/answer message. Enable this field when a gateway interoperates with the devices that support offer/answer NoAudio codec. The default value is n.

Modulation Mode

Avaya V.150.1 supports the V.32, the V.34, the V.90, and the V.92 modem modulations. For users who need other modulations, you must set the **Modem Mode** field to pass-through. The following fields show whether V.150.1 is enabled to transport the respective modulations as Modem Relay:

V.32

Valid entry	Usage
y	Use this entry to enable V.32 modem modulation. The default value is y.
n	Use this entry to disable V.32 modem modulation.

V.34

Valid entry	Usage
y	Use this entry to enable V.34 modem modulation. The default value is y.
n	Use this entry to disable V.34 modem modulation.

V.90

Valid entry	Usage
y	Use this entry to enable the V.90 digital modem modulation. The default value is y. **Note:** The system sets the **V.34** field to display-only with the default value y.
n	Use this entry to disable the V.90 digital modem modulation.

V.92

Valid entry	Usage
y	Use this entry to enable the V.92 digital modem modulation. **Note:** The system sets the **V.34** field to display-only with the default value y.
n	Use this entry to disable the V.92 digital modem modulation. The default value is n.

SPRT Retransmissions

Use this field to assign the maximum number of data retransmissions to Transport Channel 1 (TC1) and Transport Channel 2 (TC2). You can assign a value from 1 to 32. The default value is 3.

SPRT TC0 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 0. You can enter a value from 140 to 256. The default value is 140 bytes.

Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware version.

SPRT TC1 Payload Size

Use this field to assign the maximum payload size for SPRT Transport Channel 1. You can enter a value between 132 to 256. The default value is 132 bytes.

Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT TC1 Window Size

Use this field to assign the window size to SPRT Transport Channel 1. You can enter a value from 32 to 96. The default value is 32 bytes.

SPRT TC2 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 2. You can enter a value from 132 to 256. The default value is 132 bytes.

Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT TC2 Window Size

Use this field to assign the window size to SPRT Transport Channel 2. You can enter a value from 8 to 32. The default value is 8 bytes.

SPRT TC3 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 3. You can enter a value from 140 to 256. The default value is 140 bytes.

Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT Timer TC1-TA01

Use this field to set the value of the acknowledgement timer of Transport Channel 1. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 90.

SPRT Timer TC1-TA02

Use this field to set the value of the acknowledgement update timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 130.

SPRT Timer TC1-TR03

Use this field to set the value of the retransmit timer of Transport Channel 1. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SPRT Timer TC2-TA01

Use this field to set the value of the acknowledgement timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 90.

SPRT Timer TC2-TA02

Use this field to set the value of the acknowledgement update timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SPRT Timer TC2-TR03

Use this field to set the value of the retransmit timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SSE Inter-Packet Interval

Use this field to assign an interval between the State Signaling Events (SSE) packets. You can enter a value between 10 to 40 with the increments of 10. The default value is 20 mSec.

SSE Repetition Redundancy (Packets)

Use this field to assign the number of redundant SSE packets for transmission. The redundancy option provides robust transport of data packets across an IP-based network. You can enter a value from 0 to 3. The default value is 3.

V.42 Error Correction

V.42 is an error correction protocol for V.150.1 media transport between modems. If you set this field to y, the gateways use the **V.42 Error Correction** protocol. The default value is n.

Chapter 67: IP Interfaces

Assigns a network region to an IP interface device, or administers Ethernet options.

The appearance of the IP Interfaces screen can vary according to the interface type you are administering, and your system's configuration.

Note:

After starting the process of administering the IP interface for the TN2602AP circuit pack, any active calls continue to use the TN2602AP circuit pack's physical IP address for the connection, not the virtual IP address administered here. Therefore, any calls that continue after administering the virtual address, drop in the event of an interchange.

Example command: `add ip-interface n`, where *n* is the board location.

IP Interface: page 1

Allow H.248 Gateways

Controls whether branch gateways (G700, G450, G430, G350, and G250) can or cannot register on the interface.

Valid Entry	Usage
y	On a simplex main server, enables H.248 endpoint connectivity to the Processor Ethernet (PE) interface. Used for a Survivable Remote Server (Local Survivable Processor).
n	Disables H.248 endpoint connectivity to the PE interface. H.248 endpoint connectivity using the PE interface on a Survivable Core Server (Enterprise Survivable Server) is not supported.

Allow H.323 Endpoints

Controls whether H.323 endpoints can or cannot register on the interface.

Valid Entry	Usage
y	On a simplex main server, enables H.323 endpoint connectivity to the Processor Ethernet (PE) interface. Used for a Survivable Remote Server (Local Survivable Processor).
n	Disables H.323 endpoint connectivity to the PE interface. H.323 endpoint connectivity using the PE interface on a Survivable Core Server (Enterprise Survivable Server) is not supported.

Code/Sfx

Circuit pack TN code and suffix.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Related Links

Critical Reliable Bearer on page 427

Critical Reliable Bearer

Enables or disables a duplicate TN2602 circuit pack in a port. Available only with the TN2602.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Enable Ethernet Interface

Valid Entry	Usage
y	Enables the Ethernet port associated with the TN2602AP circuit pack.
n	Disables when there is no standby, or when the standby has been disabled. Should be disabled before administering the IP interface.

Ethernet Link

The administered link number for an Ethernet link.

Gateway Node Name

The gateway node name associated with the IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is enabled. Accepts up to 15 characters.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Related Links

Critical Reliable Bearer on page 427

Gatekeeper Priority

Available only if H.323 endpoints are enabled and the Communication Manager server is a main server or a Survivable Remote Server. Unavailable on a Survivable Core Server.

Valid Entry	Usage
1 to 9	Sets a priority on the interface that affects where the interface is displayed on the gatekeeper list. The value in this field is used on the alternate gatekeeper list. The lower the number, the higher the priority. Default is 5.

Related Links

Allow H.323 Endpoints on page 426

Network Region

Valid Entry	Usage
1 to 2000	The value of the network region where the TN2602AP resides. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is enabled. **Note:** The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Related Links

Critical Reliable Bearer on page 427

Network uses 1's for Broadnet Addresses

Allows or denies using a broadcast address to send the same message to all systems or clients on a local area network.

Node Name

The node name associated with the IP address of the TN2602AP circuit pack. Accepts up to 15 characters.

Related Links

Near-end Node Name on page 682
Name on page 451
Group Type on page 675

Receive Buffer TCP Window Size

Valid Entry	Usage
512 to 8320	The number of bytes allotted for the buffer that receives TCP data for a TN799 (CLAN) circuit pack. The default is 512.

Slot

The slot location. Requires entry of the location of the second TN2602AP circuit pack for a non-duplicated board.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth character are the slot number.

Related Links

Critical Reliable Bearer on page 427

Subnet Mask

A 32-bit binary number that divides the network ID and the host ID in an IP address. This is the subnet mask for TN2602AP. Also applies to the second TN2602AP circuit pack when the critical reliable bearer is enabled.

Related Links

Critical Reliable Bearer on page 427

Target socket load

The maximum number of sockets targeted for this interface. The default is 80% of the platform maximum. Used for load balancing endpoint traffic across multiple IP interfaces. Controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number allocated, the system continues to add sockets until the interface is at its maximum capacity. Available only with a procr type IP interface.

Note:

The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid Entry	Usage
1 to 3500	S8510, S8800, and duplex servers

Related Links

Type on page 431

Target socket load and Warning level

Controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number allocated, the system continues to add sockets until the interface is at its maximum capacity. If the targeted percentage is exceeded on a CLAN, a warning alarm is generated.

If there is only one IP interface within a priority, the target socket load and warning level is not used for load balancing. A value in this field can be used to receive an error or a warning alarm if the targeted value is exceeded. Available only with CLAN type IP interfaces.

Note:

The 4606, 4612, and 4624 telephones do not support the load balancing feature of the TN2602AP circuit pack.

Valid Entry	Usage
1 to 499	The maximum number of sockets targeted for this interface. If the number of sockets exceeds the targeted number, a warning alarm is generated. The default is 400.

Related Links

Type on page 431

Type

The type of IP interface.

Valid Entry	Usage
clan	Control Local Area Network (C-LAN) board
VAL	Voice Announcement LAN board
medpro	Media Processor board
procr	Processor — S8300D S8300E S8800, and duplex media servers

VLAN

Sends Virtual Local Area Network (VLAN) instructions to the PROCR (S8300D/duplex Media Servers), CLAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. Unavailable for Voice Announcement over LAN (VAL) boards.

Valid Entry	Usage
0 to 4095	Specifies the virtual LAN value.
n	Disabled. This is the default.

IP Interface: page 2

ETHERNET OPTIONS

The Ethernet port associated with the TN2602AP must be disabled before any changes can be made to these fields.

Auto?

Valid Entry	Usage
y	Enables auto-negotiation. Default.
n	Applies manual speed and duplex settings.

Duplex

Available only if auto-negotiation is disabled.

Valid Entry	Usage
Full	The full duplex setting for this IP board. Default when speed is set to 100 Mbps.
Half	The half duplex setting for this IP board. Default.

Related Links

Auto? on page 432

Speed

Available only if auto-negotiation is disabled.

Valid Entry	Usage
10Mbps	The speed of the Ethernet connection.
100Mbps	The only speed option available for the TN2602AP circuit pack is 100Mbps. This is the default and cannot be changed.

Related Links

Auto? on page 432

IPV6 PARAMETERS

Enable Ethernet Interface

Valid Entry	Usage
y	Enables the Ethernet port associated with the TN2602AP circuit pack.
n	Disables when there is no standby, or when the standby has been disabled. Should be disabled before administering the IP interface.

Ethernet Link

The administered link number for an Ethernet link.

Gateway Node Name

The gateway node name associated with the IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is enabled. Accepts up to 15 characters.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Related Links

Critical Reliable Bearer on page 427

Node Name

The node name associated with the IP address of the TN2602AP circuit pack. Accepts up to 15 characters.

Related Links

Near-end Node Name on page 682
Name on page 451
Group Type on page 675

Subnet Mask

A 64-bit binary number that divides the network ID and the host ID in an IP address. This is the subnet mask for TN2602AP. Also applies to the second TN2602AP circuit pack when the critical reliable bearer is enabled.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from active to standby media processor is in process, calls might be dropped.

Related Links

Critical Reliable Bearer on page 427

IP Interfaces: page 3

VOIP/NETWORK THRESHOLDS

Enable VoIP/Network Thresholds

Enables or disables the recording of Voice/Network Statistics at a system level for a single media processor board. This applies to both TN2602 boards, if duplicated. Any changes to the value of this field, results in an updated message sent to the media processor board.

Jitter (ms)

Available only if VoIP/Network thresholds are enabled and the board type is a media processor.

Valid Entry	Usage
0 to 9999	The unacceptable jitter coming into the media processor board at which point data is captured and sent to Communication Manager. Default is 50 milliseconds.

Related Links

Enable VoIP/Network Thresholds on page 434
Type on page 431

Packet loss (%)

Available only if VoIP/Network thresholds are enabled and the board type is a media processor.

Valid Entry	Usage
0 to 100	The percentage of the unacceptable packet loss coming into the administered media processor board. Default is 5. **Note:** xxx indicates 100% packet loss.

Related Links

Enable VoIP/Network Thresholds on page 434
Type on page 431

RT Delay (ms)

Available only if VoIP/Network thresholds are enabled and the board type is a media processor.

Valid Entry	Usage
1 to 9999	Round Trip Delay is the unacceptable elapsed time for a packet to reach a remote location and return. Default is 500 milliseconds.

Related Links

Enable VoIP/Network Thresholds on page 434
Type on page 431

Chapter 68: IP network region

Configures within-region and between-region connectivity settings for all VoIP resources and endpoints within a given IP region. The first page is used to modify the audio and QoS settings. The **Codec Set** field on this page reflects the CODEC set that must be used for connections between telephones within this region or between telephones and MedPro/Prowler boards and gateways within this region. The ability to do NAT shuffling for direct IP-to-IP audio connections is also supported.

Example command: `change ip-network-region` *n*, where *n* is the network region number.

IP network region: page 1

Authoritative Domain

The domain for which this network region is responsible. The system displays the Authoritative Domain in the **From** header of any SIP messages. Accepts a name or IP address consisting of up to 20 characters.

Location

Note:

If the Multinational Locations feature is enabled, and IP telephones derive their network region from the IP Network Map, administer this field with a valid value (1 to 2000). With this, the IP endpoints can use the right VoIP resources.

Valid Entry	Usage
1 to 250 or 1 to 2000 Depending on the server configuration. For more information, see *Avaya Aura® Communication Manager*	Assigns the location number to the IP network region. Correct date and time information, and trunk routing based on IP network region can be configured. The IP endpoint uses this as its location number. See the Location sections in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205, for the other ways to assign a location, and for a list of features that use location.

Table continues…

Valid Entry	Usage
System Capacities Table, 03-300511.	
blank	Obtains the location from the cabinet containing the CLAN or the gateway where the endpoint is registered. By default, the value is blank.

Name

A description of the region. Accepts up to 20 characters.

Region

The number of the network region being administered.

Stub Network Region

Using this field, you can create a core network region or a stub network region.

Valid entry	Usage
y	Creates a stub network region. In the **dst rgn** field, you must enter the number of the destination network region that directly connects with this stub network region. **Note:** If you are converting a core network region to a stub network region, you must ensure that the core network region is directly connected with only one core network region. A stub network must have only one direct connection with a core network region.
n	Creates a core network region. For network regions 1 to 250, the default value of this field is n. For network regions 251 to 2000, this field is noneditable and displays a default of y.

MEDIA PARAMETERS

Codec Set

Valid Entry	Usage
1 to 7	The number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a network region.

Valid Entry	Usage
y	Saves on bandwidth resources and improves sound quality of voice over IP transmissions.
n	Might be used if, for example, the IP telephones within the region are behind two or more firewalls.
native(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone/ softphone itself (without being translated by NAT). IP telephones must be configured behind a NAT device before this entry is enabled.
translated(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device before this entry is enabled.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions

Valid Entry	Usage
y	Saves on bandwidth resources and improves sound quality of voice over IP transmissions.
n	Might be used if, for example, the IP telephones within the region are behind two or more firewalls.
native(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP telephones must be configured behind a NAT device before this entry is enabled.
translated(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device before this entry is enabled.

IP Audio Hairpinning

If enabled, IP endpoints connect through the IP circuit pack in the server in IP format to bypass the Communication Manager TDM bus.

UDP PORT RANGE

UDP Port Range Min

Valid Entry	Usage
1024 to 65534	The minimum range of the UDP port number used for audio transport. Defaults to 2048 to 3028.

UDP Port Range Max

Valid Entry	Usage
1025 to 65535	The maximum range of the UDP port number used for audio transport. Defaults to 2048 to 3028.

DIFFSERV/TOS PARAMETERS

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop.

Valid Entry	Usage
0 to 63	The decimal equivalent of the DiffServ Audio PHB value. Default is 46.

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop.

Valid Entry	Usage
0 to 63	The decimal equivalent of the Call Control PHB value. Default is 34.

Video PHB Value

Valid Entry	Usage
0 to 63	The decimal equivalent of the DiffServ Video PHB value. Default is 26.

802.1P/Q PARAMETERS

Audio 802.1p Priority

Valid Entry	Usage
0 to 7	Provides Layer 2 priority for Layer 2 switches. Changes take effect after circuit pack reset, telephone reboot, or system reset.

Call Control 802.1p Priority

Valid Entry	Usage
0 to 7	Provides Layer 2 priority for Layer 2 switches. Changes take effect after circuit pack reset, telephone reboot, or system reset.

Video 802.1p Priority

Valid Entry	Usage
0 to 7	The Video 802.1p priority value. Changes take effect after circuit pack reset, telephone reboot, or system reset.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

Enables or disables retries when RSVP fails.

Available only if RSVP is enabled.

Related Links

RSVP Enabled on page 440

RSVP Enabled

Enables or disables RSVP.

RSVP Profile

Available only if RSVP is enabled.

Set this field to what you have configured on your network.

Valid Entry	Usage
guaranteed-service	Limits end-to-end queuing delay from sender to receiver. This setting is best for VoIP applications.

Table continues…

Valid Entry	Usage
controlled-load	This subset of guaranteed-service provides for a traffic specifier, but not end-to-end queuing delay.

RSVP Refresh Rate (secs)

Available only if RSVP is enabled.

The system displays this field only if the **RSVP Enabled** field is set to y.

Valid Entry	Usage
1 to 99	The RSVP refresh rate in seconds.

Related Links

RSVP Enabled on page 440

RSVP unreserved (BBE) PHB Value

Valid Entry	Usage
0 to 63	The BBE codepoint is used whenever an RSVP reservation is being obtained (pending), or has failed in some way, to provide efficient service to the voice stream.

H.323 IP ENDPOINTS

H.323 Link Bounce Recovery

Enables or disables the H.323 Link Bounce Recovery feature for this network region. The default is enabled.

Idle Traffic Interval (seconds)

Valid Entry	Usage
5 to 7200	The maximum traffic idle time in seconds after which a TCP Keep-Alive (KA) signal is sent from the endpoint.. Default is 20.

Keep-Alive Interval (seconds)

Valid Entry	Usage
1 to 120	Sets the interval between TCP Keep-Alive re-transmissions. When no ACK is received for all retry attempts, the local TCP stack ends the TCP session and the associated socket is closed. Default is 5.

Keep-Alive Count

Valid Entry	Usage
1 to 20	Sets the number of times the Keep-Alive message is transmitted if no ACK is received from the peer. Default is 5.

IP network region: page 2

RTCP Reporting Enabled

If enabled, sends RTCP Reports to a special server, such as for the VMON tool.

Note:

Regardless of how this field is administered, RTCP packets are always sent peer-to-peer.

RTCP MONITOR SERVER PARAMETERS

IPV4 Server Port

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
1 to 65535	The port for the RTCP Monitor server. Default is 5005.

Related Links

Use Default Server Parameters on page 443
RTCP Reporting Enabled on page 442

IPV6 Server Port

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
1 to 65535	The port for the RTCP Monitor server. Default is 5005.

Related Links

Use Default Server Parameters on page 443
RTCP Reporting Enabled on page 442

RTCP Report Period (secs)

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
5 to 30	The report period for the RTCP Monitor server in seconds.

Related Links

Use Default Server Parameters on page 443

Server IPV4 Address

The IPv4 address for the RTCP Monitor server.

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Related Links

RTCP Reporting Enabled on page 442
Use Default Server Parameters on page 443

Server IPV6 Address

The IPv6 address for the RTCP Monitor server.

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Related Links

RTCP Reporting Enabled on page 442
Use Default Server Parameters on page 443

Use Default Server Parameters

If enabled, uses the system-wide default RTCP Monitor server parameters. Available only if RTCP Reporting is enabled.

Related Links

RTCP Reporting Enabled on page 442

IP network region: page 3

This page covers the information for Inter-Gatewày Alternate Routing (IGAR), backup server names in priority order, and security procedures.

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY

If **Inter-Gateway Alternate Routing** (IGAR) is enabled for any row on subsequent pages, the following fields for each network region must be administered to route the bearer portion of an IGAR call.

Conversion to Full Public Number - Delete

Valid Entry	Usage
0 to 7	The digits to delete

Conversion to Full Public Number - Insert

Valid Entry	Usage
0 to 13 blank	The number of digits to insert. International numbers should begin with plus (+). The Inter-Gateway Alternate Routing (IGAR) and Dial Plan Transparency (DPT) features convert the plus (+) digit to appropriate international access code when starting the trunk call. **Note:** The optional plus (+) at the beginning of the inserted digits is an international convention indicating that the local international access code must be dialed before the number.

Dial Plan Transparency in Survivable Mode

Valid Entry	Usage
y	Enables the Dial Plan Transparency feature when a gateway registers with a Survivable Remote Server (Local survivable processor), or when a port network registers with a Survivable Core Server (Enterprise Survivable Server).
n	Default is n.

Incoming LDN Extension

An extension used to assign an unused Listed Directory Number for incoming IGAR calls.

Maximum Number of Trunks to Use for IGAR

It is necessary to impose a limit on the trunk usage in a particular port network in a network region when Inter-Gateway Alternate Routing (IGAR) is active. The limit is required because if there is a major IP WAN network failure, it is possible to use all trunks in the network region(s) for IGAR calls.

Valid Entry	Usage
1 to 999, or blank	The maximum number of trunks to be used for Inter-gateway alternate routing (IGAR).

BACKUP SERVERS IN PRIORITY ORDER

Lists the backup server names in priority order. Backup server names should include Survivable Remote Server names and Survivable Core Server names. If you are using the Processor Ethernet, the backup servers list must include the survivable core PE address else the phones will not register to the survivable core during a failure. Any valid node name is a valid entry. Valid node names can include names of Customer LANs, ICCs, Survivable Core Servers, and Survivable Remote Servers.

H.323 SECURITY PROFILES

Permitted security profiles for endpoint registration in the network region. You must enter at least one security profile. Otherwise, no endpoint will be permitted to register from the region.

Valid Entry	Usage
challenge	Includes the various methods of PIN-based challenge and response schemes in current use. This is a relatively weak security profile.
pin-eke	The H.235 Annex H SP1.
strong	Permits the use of any strong security profile. The `H323TLS` profile is the strongest security profile in Communication Manager.
any-auth	Includes any of the security profiles.
H323TLS	Communication Manager apples this security profile when the network region of an H.323 phone is administered with H323TLS or Strong security profiles. Also, Communication Manager and the endpoint negotiate by using the H323 TLS profile. H323TLS profile sends H.323 signaling messages through a TLS-encrypted channel.

Allow SIP URI Conversion

Administers whether or not a SIP URI should be permitted to change. Degrading the URI from sips//: to sip//: might result in a less secure call. This is required when SIP SRTP endpoints are allowed to make and receive calls from endpoints that do not support SRTP.

Valid Entry	Usage
y	Allows conversion of SIP URIs. Default is y.
n	No URI conversion. Calls from SIP endpoints that support SRTP made to other SIP endpoints that do not support SRTP will fail. However, if you enter y for the **Enforce SIPS URI for SRTP** field on the signaling group screen, URI conversion takes place independent of the value set for the **Allow SIP URI conversion** field on the IP Network Region screen.

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS

Near End Establishes TCP Signaling Socket

Indicates whether Communication Manager (the near end) can establish the TCP socket for H.323 IP endpoints in this network region.

Valid Entry	Usage
y	Communication Manager determines when to establish the TCP socket with the IP endpoints, assuming the endpoints support this capability. This is the default.
n	The IP endpoints always attempt to set up the TCP socket immediately after registration. This field should be disabled only in network regions where a non-standard H.323 proxy device or a non-supported network address translation (NAT) device would prevent the server from establishing TCP sockets with H.323 IP endpoints.

Near End TCP Port Min

Valid Entry	Usage
1024 to 65531	The minimum port value used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1). Default is 61440.

Related Links

Near End TCP Port Max on page 446

Near End TCP Port Max

Valid Entry	Usage
1028 to 65535	The maximum port value to be used by the Control Lan (C-LAN) circuit pack or processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1). Default is 61444.

Related Links

Near End TCP Port Min on page 446

IP network region: page 4

Each subsequent page shows the inter-region connectivity for 15 region pairs.

AGL

The maximum number of destination region IP interfaces included in alternate gatekeeper lists (AGL).

Valid Entry	Usage
0 to 16	Communication Manager uses the numeric value of gatekeeper addresses.
all	Communication Manager includes all possible gatekeeper addresses in the endpoint's own network region and in any regions to which the endpoint's region is directly connected.
blank	The administration field is ignored.

codec-set

Valid Entry	Usage
1 to 7 pstn blank	The codec set used between the two regions. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions. If the two regions are disconnected at all, this field should be blank.

direct-WAN

Indicates whether the two regions (source and destination) are directly connected by a WAN link. The default value is enabled if a **codec-set** is administered.

Related Links

codec-set on page 447

dst rgn

Valid Entry	Usage
1 to 250	The destination region for this inter-network connection.

Dyn CAC

Available only if the **WAN-BW-limits (Units)** is Dynamic. The gateway must be configured to be a CAC (Call Admission Control) gateway.

Valid Entry	Usage
1 to 250 blank	The gateway that reports the bandwidth-limit for this link. Default is blank.

Related Links

WAN-BW-limits (Units) on page 450

IGAR

Allows pair-wise configuration of Inter-Gateway Alternate Routing (IGAR) between network regions.

Valid Entry	Usage
y	Enables IGAR capability between this network region pair. Default for a pstn codec set.
n	Disable IGAR capability between this network region pair. Default, except for a pstn codec set.
f	Forced. Moves all traffic onto the PSTN. This option can be used during initial installation to verify the alternative PSTN facility selected for a network region pair. This option can also be used to temporarily move traffic off of the IP WAN if an edge router is having problems or an edge router needs to be replaced between a network region pair.

Intervening-regions

Allows entry of intervening region numbers between the two indirectly-connected regions.

Valid Entry	Usage
1 to 250	Up to four intervening region numbers between the two indirectly-connected regions. **Note:** Indirect region paths cannot be entered until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

Mtce

Valid Entry	Usage
t	This is a test-only option. Inter-region connectivity testing is performed for the network region pair by using a simple PING sent between entities in each network region. If a test fails, only an error is added to the system error log. IP media connections

Table continues…

Valid Entry	Usage
	between the region pair are never blocked. The testing is done at the rate of not more than once per 5 minutes.
m	This is a measurement based option. Inter-region connectivity testing is performed by a continuous set of PINGs sent between entities in each network region. The Ping Test Interval (sec) and Number of Pings Per Measurement Interval fields control the rate of testing. The Roundtrip Propagation Delay (ms) and Packet Loss (%) thresholds control success or failure. If the, test measurements exceed the administered thresholds; future IP media connections between the network region pair will be blocked.
d	No testing is performed for the network region pair.

Related Links

Ping Test Interval (sec) on page 454
Number of Pings Per Measurement Interval on page 454
Roundtrip Propagation Delay (ms) on page 455
Packet Loss (%) on page 454

src rgn

Valid Entry	Usage
1 to 250	The source region for this inter-network connection.

Sync

The system displays Sync when the **Synchronization over IP** field is enabled.

Valid Entry	Usage
y	Timing IGC streams are allowed between the region pair that is being administered. The default value is y.
n	Do not allow timing IGC streams between the region pair that is being administered.

Video (Norm)

Valid Entry	Usage
0 to 9999 for Kbits 0 to 65 for Mbits blank for NoLimit	The amount of bandwidth to allocate for the normal video pool to each IP network region.

Video (Prio)

Valid Entry	Usage
0 to 9999 for Kbits 0 to 65 for Mbits blank for NoLimit	The amount of bandwidth to allocate for the priority video pool to each IP network region.

Video (Shr)

Specifies whether the normal video pool can be shared for each link between IP network regions.

WAN-BW limits (Total)

Valid Entry	Usage
1 to 9999	The bandwidth limit for direct WAN links. Values for this field can be entered in the number of connections, bandwidth in Kbits or calls, or left blank for NoLimit.
1 to 65	Values for this field can be entered in the number of connections, bandwidth in Mbits, or left blank for NoLimit.

WAN-BW-limits (Units)

Valid Entry	Usage
Calls Dynamic Kbits/sec Mbits/sec blank for NoLimit	The unit of measure corresponding to the value entered for bandwidth limitation. Bandwidth should be limited by the number of connections, bandwidth in Kbits/sec, or bandwidth in Mbits/sec, or left blank. Default is blank.

Chapter 69: IP Node Names

Administers node names and IP addresses for the switch and the terminal server media processors administered on the IP Interfaces screen.

Note:

The system displays the Processor Ethernet interface node name (procr or procr6) on the IP Node Names screen. The PE interface node name cannot be added to the IP Node Names screen. The line containing the keywords procr or procr6 displays the IP address.

Example command: `change node-names ip`

Name

The name of the adjunct, server, or switch node used as a label for the associated IP address. The node names must be unique for each server or switch. Uses up to 15 alpha-numeric characters.

IP Address

A unique IP address assigned to each port on any IP device that is used for a connection. Also supports IPv6 address format.

Chapter 70: IP options system parameters

Example command: `change system-parameters ip-options`

IP options system parameters: page 1

AUTOMATIC TRACE ROUTE ON

Link Failure

Enables or disables the automatic trace route command. If enabled, to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its port networks, gateways, or IP trunks is lost.

Note:

If disabled, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are started or logged. In other words, the link failure buffer is cleared.

H.248 MEDIA GATEWAY

Link Loss Delay Timeout (minutes)

Valid Entry	Usage
1 to 30	The number of minutes to delay the reaction of the call controller to a link bounce. Assists with the H.248 link bounce recovery mechanism of the Avaya G700 branch gateway. Specifically, prevents the call controller from removing all boards and ports prematurely in response to a link bounce. Default is 5.

H.323 IP ENDPOINT

Link Loss Delay Timer (minutes)

Valid Entry	Usage
1 to 60	The number of minutes to delay the reaction of the call controller to a link bounce. Specifies how long the Communication Manager server preserves registration and any stable calls that might exist on the endpoint after it has lost the call signaling channel to the endpoint. If the endpoint does not re-establish connection within this period, the system cancels the registration and any calls of the endpoint. This timer does not apply to soft IP endpoints operating in telecommuter mode. Default is 5.

Periodic Registration Timer (min)

This timer is started when an IP telephone registration is taken over by another IP endpoint. When the timer expires, the telephone tries to reregister with the server. Default timer value is dependent on the number of unsuccessful periodic registration attempts. Sample field values apply unless the endpoint is interrupted, such as by power loss, or the user takes manual action to override this automatic process:

- 20 means once every 20 minutes for two hours, then once an hour for 24 hours, then once every 24 hours continually.
- 60 means once an hour for two hours, then once an hour for 24 hours, then once every 24 hours continually.

Valid Entry	Usage
1 to 60	The number of minutes before an IP telephone registration is taken over by another IP endpoint attempts to re-register with the server. Default is 60.

Primary Search Time (seconds)

Valid Entry	Usage
15 to 3600	The maximum number of seconds the IP endpoint attempts to register with its current Communication Manager server while the telephone is disconnected before going to a Survivable Remote Server. With this timer, the customer can specify the maximum time that an IP endpoint spends on trying to connect to the C-LANS before going to a Survivable Remote Server. When the IP telephone's receiver is lifted, the endpoint continues trying to re-establish connection with the current server until the call ends. Default is 75.

Short/Prefixed Registration Allowed

The system displays Short/Prefixed Registration Allowed if the **IP Stations** field on the System Parameters Customer Options screen is set to y.

Valid Entry	Usage
y	Call Processing allows an IP endpoint to register using a short extension, for the extensions that have **Short/Prefixed Registration Allowed** field set to default on the Station screen. The default value is y.
n	Call Processing does not allow an IP endpoint to register using a short extension, for the extensions that have **Short/Prefixed Registration Allowed** field set to default on the Station screen.

Related Links

Short/Prefixed Registration Allowed on page 738

IP MEDIA PACKET PERFORMANCE THRESHOLDS

Enable Voice/Network Stats

Enables or disables the recording of voice and network statistics at a system level for all TN2302/ TN2602 media processor boards in the network. The default value is disabled.

Number of Pings Per Measurement Interval

Valid Entry	Usage
10 to 100	the number of test pings that comprise a measurement from which the performance values (delay and loss) are calculated. Default is 10.

Packet Loss (%)

Specifies thresholds to be applied to packet loss rates (as measured by ping) for determining activation or deactivation of signaling group bypass.

High:

Valid Entry	Usage
0 to 100	The high value cannot be less than the minimum value. Default is 40.

Low:

Valid Entry	Usage
0 to 100	The low value cannot be more than the maximum value. Default is 15.

Ping Test Interval (sec)

Valid Entry	Usage
10 to 999	The time between performance test pings for each testable signaling group. Default is 20.

Roundtrip Propagation Delay (ms)

Specifies thresholds to be applied to roundtrip packet propagation delays as measured by ping, for use in activating or clearing signaling group bypass.

High:

Valid Entry	Usage
10 to 9999	The high value cannot be less than the minimum value. Default is 800.

Low:

Valid Entry	Usage
10 to 9999	The low value cannot be more than the maximum value. Default is 400.

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS

Announcement Server IP Address

The IP address of the Announcement Server that is a unique IP address assigned to each port on any IP device that is used for a connection.

Announcement Storage Path Name

The directory path name on the Announcement Server where the announcements are stored. Accepts up to 40 characters.

Login

The login used by the gateway to access the announcement server. Accepts up to 10 characters.

Password

The password used by the gateway to access the announcement server. Accepts up to 10 characters.

RTCP MONITOR SERVER

The RTCP monitor is a separate computer that receives RTCP packets from many devices. Communication Manager pushes these values to IP telephones, IP softphones and VoIP media modules, such that they know where to send the data.

Default RTCP Report Period (secs)

Valid Entry	Usage
5 to 99	The number of seconds IP telephones, IP softphones, and VoIP media modules send RTCP packets to the RTCP server.

Default Server IP Address

The default IP address of the RTCP server used for each administered region. A unique IP address is assigned to each port on any IP device that is used for a connection.

Related Links

Server IPV4 Address on page 443

Default Server Port

Valid Entry	Usage
1 to 65535	The default TCP/IP port of the RTCP server. Default is 5005.

Link Failure

Enables or disables the automatic trace route command. If enabled, to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its port networks, gateways, or IP trunks is lost.

Note:

If disabled, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are started or logged. In other words, the link failure buffer is cleared.

IP options system parameters: page 2

Prefer use of G.711 by Music Sources

Enables or disables using G.711 for intra-switch Music-On-Hold. The default value is n.

Force Phones and Gateways to Active LSPs

Enables or disables forcing telephones and gateways to active LSPs. The default is disabled.

Override ip-codec-set for SIP direct-media connections

When you enable **Override ip-codec-set for SIP direct-media connections**, the system overrides Communication Manager ip-codec-set preferences and uses SIP endpoint preferences.

Valid entry	Usage
n	SIP endpoints with direct-media connection negotiate media streams according to the ip-codec-set preferences. Communication Manager filters and reorders the audio codecs present in the SDP offer that is part of the SIP INVITE message. The default value is n.
y	SIP endpoints with direct-media connection negotiate media streams according to the endpoint preferences. Communication Manager does not filter or reorder audio codecs present in the SDP offer that is part of the SIP INVITE message as long as available bandwidth is sufficient for all codecs present in the offer. If the available bandwidth is not sufficient, Communication Manager retains the codec ordering present in the SDP offer but removes the codecs that do not fit in the available bandwidth.

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS

Enable Detection and Alarms

Enables or disables the hyperactive gateway registration feature. Default is disabled.

Number of Registrations within the Window

Valid Entry	Usage
1 to 19	The number of registrations that occur within the hyperactivity window for generating a Gateway alarm. Default is 3. Available only if detection and alarming is enabled .

Related Links

Enable Detection and Alarms on page 457

Parameters for Media Gateway Alarms: Hyperactive Registration Window (minutes)

Available only if detection and alarming is enabled.

Valid Entry	Usage
1 to 15	The time in minutes for checking hyperactive gateway registrations. Default is 4 minutes.

Related Links

Enable Detection and Alarms on page 457

Parameters for Network Region Registration (NR-REG) Alarms: % of Gateways in Network Region with Hyperactive Registration Alarms

Available only if detection and alarming is enabled.

Valid Entry	Usage
1 to 99	The percent of Gateways within an ip-network region that should be alarmed before an IP-Registration alarm is generated. Default is 80%.

Related Links

Enable Detection and Alarms on page 457

IP DTMF TRANSMISSION MODE

Inter-System IP DTMF Transmission Mode

Specifies the touchtone signals that are used for dual-tone multifrequency (DTMF) telephone signaling.

Valid Entry	Usage
in-band	All G711 and G729 calls pass DTMF in-band. DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band. It also supports the SIP trunks.
in-band-g711	Only G711 calls pass DTMF in-band. The system displays the **in-band-g711** option if the **Group Type** field is set to h.323.
out-of-band	All IP calls pass DTMF out-of-band. For IP trunks, the digits are done with either Keypad IEs or H245 indications. This is the default for newly added H.323 signaling groups. The out-of-band option enables out of band signaling for the SIP Signaling group. When you select the out-of-band option, Communication Manager sends all outgoing DTMF messages as SIP INFO messages over a SIP signaling group. This option is interoperable with H.323 networks. You can connect an Avaya non-SIP endpoint or trunk to a voice mail system on the H.323 network that is integrated with Session Manager and Communication Manager through SIP. Examples of Avaya non-SIP endpoints or trunks are H.323, Analog, DCP, and Digital trunks.
rtp-payload	This is the method specified by RFC 2833. This is the default value for newly added SIP signaling groups. Support for SIP trunks requires the default entry of rtp-payload.

Intra-System IP DTMF Transmission Mode

The IP transmission mode.

Valid Entry	Usage
in-band	DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band.
rtp-payload	Support for SIP trunks requires the entry of rtp-payload.

IP options system parameters: page 3

Use this screen to administer the SNMP station parameters and the services dialpad parameters. You can administer the parameters on the following telephone types: 4601, 4602, 4610, 4620, 4621, 4622, 4625, 96xx, and 16xx.

SERVICES DIALPAD PARAMETERS

To disable the services dialpad parameters on the telephones, set the **Download Flag** field to y and leave the **Password** field blank.

Download Flag

If you set this field to y, Communication Manager downloads the services dialpad parameters and the associated IP addresses to the telephones. The default value is n.

Password

Use this field to enter the password that a technician uses to administer craft procedures. During the installation or after the successful installation of an IP telephone, the technician performs craft procedures to customize the telephone installation for specific operating environment. You can provide a password of length up to seven digits or leave the field blank. The default value is 27238.

SNMP STATION PARAMETERS

To disable the SNMP parameters on the telephones, set the **Download Flag** field to y and leave the **Community String** field blank.

Community String

Use this field to enter a string that an IP telephone uses to determine whether the telephone must receive SNMP messages. You can enter up to 32 characters.

The telephone does not respond to the incoming SNMP message if you:

- leave the **Community String** field blank and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**

- leave the **Community String** field blank and the source address of the SNMP message matches with an address in the **SNMP Source Address list**
- set the **Community String** field to a value and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
-

The telephone responds to the incoming SNMP message only if you set the **Community String** field to a value and the source address of the SNMP message matches with an address in the **SNMP Source Address list**.

Download Flag

If you set this field to y, Communication Manager downloads the SNMP parameters and the associated IP addresses to the telephones. The default value is n.

SOURCE ADDRESSES

Use this field to enter node names, mapping to proper IP address, that an IP telephone uses to validate the source address of an SNMP message. The telephone does not respond to the incoming SNMP message if you:

- leave the **Community String** field blank and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
- leave the **Community String** field blank and the source address of the SNMP message matches with an address in the **SNMP Source Address list**
- set the **Community String** field to a value and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
-

The telephone responds to the incoming SNMP message only if you set the **Community String** field to a value and the source address of the SNMP message matches with an address in the **SNMP Source Address list**.

You can enter up to six node names. Use the IP Node Names screen to administer node names.

Related Links

IP Address on page 451

IP options system parameters: page 4

Dest # 1, 2, or 3 IP address

The valid destination IPv4 address. The default destination address is 0.0.0.0.

Local Facility

Valid Entry	Usage
local0 to local7	Displays the help message upon acceptable values for local use. The default value is local4.

Port

Valid Entry	Usage
1 to 65535	The valid port number associated with the destination IPv4 address. The default port number is 514.

Related Links

Dest # 1, 2, or 3 IP address on page 461

MUSIC/ANNOUNCEMENTS IP-CODEC PREFERENCES

In general, when operating across the WAN with limited bandwidth facilities, the ip-codec-set is configured only with the compressed voice codecs. Sometimes it is necessary to carry the voice calls with the normal configured compressed codecs, and the music or announcement sources are received with non-compressed G.711 codecs.

If you administer the following fields, when possible, the system overrides the ip-codec-set preference which can be configured to prefer a compressed codec, with non-compressed G.711. If the device receives the music or announcement source which has signaled support for G.711, the system attempts to use G.711.

Prefer use of G.711 by Announcement Sources

Overrides the ip-codec-set preference and establishes inter-PN or inter-gateway connections transmitting announcements with G.711. The default value is n.

Prefer use of G.711 by IP Endpoints Listening to Announcements

Overrides the ip-codec-set preference and reconfigures the IP endpoints listening to an announcement source with G.711. The default value is n.

Prefer use of G.711 by IP Endpoints Listening to Music

Overrides the ip-codec-set preference and reconfigures the IP endpoints listening to a music source with G.711. The default value is n.

Prefer use of G.711 by Music Sources

Overrides the ip-codec-set preference and establishes inter-PN or inter-gateway connections transmitting music with G.711. The default value is n.

Chapter 71: IP Routing

There is one-to-one mapping between the **Network Bits** and the **Subnet Mask** fields; entering a value in one field uniquely determines the other field. A list of Subnet Mask addresses and their corresponding Network Bits are shown in Table below.

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	/ 0
1	128.0.0.0	2,147,483,646	/ 1
2	192.0.0.0	1,073,741,822	/ 2
3	224.0.0.0	536,870,910	/ 3
4	240.0.0.0	268,435,454	/ 4
5	248.0.0.0	134,217,726	/ 5
6	252.0.0.0	67,108,862	/ 6
7	254.0.0.0	33,554,430	/ 7
8	255.0.0.0	16,777,214	/ 8
9	255.128.0.0	8,388,606	/ 9
10	255.192.0.0	4,194,302	/ 10
11	255.224.0.0	2,097,150	/ 11
12	255.240.0.0	1,048,574	/ 12
13	255.248.0.0	524,286	/ 13
14	255.252.0.0	262,142	/ 14
15	255.254.0.0	131,070	/ 15
16	255.255.0.0	65,534	/ 16
17	255.255.128.0	32,766	/ 17
18	255.255.192.0	16,382	/ 18
19	255.255.224.0	8,190	/ 19
20	255.255.240.0	4,094	/ 20
21	255.255.248.0	2,046	/ 21
22	255.255.252.0	1,022	/ 22
23	255.255.254.0	510	/ 23
24	255.255.255.0	254	/ 24

Table continues…

Network Bits	Subnet Mask	Number of Hosts	Network Type
25	255.255.255.128	126	/ 25
26	255.255.255.192	62	/ 26
27	255.255.255.224	30	/ 27
28	255.255.255.240	14	/ 28
29	255.255.255.248	6	/ 29
30	255.255.255.252	2	/ 30
31	255.255.255.254	1	/31
	255.255.255.255	0	/32

Example command: `add ip-route next`

Board

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01–31	Six and seventh characters are the circuit number
1 to 250	Gateway
V1 to V9	Module

Destination Node

The name of the final destination node of the IP route for this connection.

Related Links

IP Node Names on page 451

Gateway

The node name of the first intermediate node consisting of a port on the CLAN circuit pack or a Destination Node on another IP route. If there are one or more intermediate nodes, the first

intermediate node is the Gateway. If there are no intermediate nodes between the local and remote CLAN ports for this connection, the Gateway is the local CLAN port.

Related Links

IP Node Names on page 451

Metric

Valid Entry	Usage
0	A server that has only one CLAN circuit pack installed.
1	A server that has more than one CLAN circuit pack installed.

Network Bits

A 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid Entry	Usage
0 to 32	The number of Network Bits associated with this IP route.

Route Number

Valid entry	Usage
digits	The IP route number. The upper limit can change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Chapter 72: IP Server Interface (IPSI) Administration

Adds a TN2312 IPSI (IP Server Interface) circuit pack. Uses the IP Server Interface (IPSI) to control port networks and provide tone, clock, and call classification services. The IPSI board connects to the control network by way of Ethernet.

In Communication Manager Release 5.2, the IP server interface administration for the TN2312 IPSI or the TN8412 SIPI provides support for Communication Manager—based SAT administration of IPSI Quality of Service (QoS) and Ethernet interface settings parameters. All further references to IPSI also apply to the TN8412 SIPI.

Note:

Initial IPSI settings must be done using the IPSI CLI interface.

Example command: `change ipserver-interface` *n*, where *n* is the assigned IPSI board location.

IP Server Interface (IPSI) Administration: page 1

Administer secondary ip server interface board

If enabled, assigns a secondary IPSI board.

Enable QoS

If enabled, turns on quality of service (QoS) from the server to the IPSI link. If you enable QoS for the control network, also enable it from the Web interface.

Encryption

Enables socket encryption for the server and IPSI link.

Ignore Connectivity in Server Arbitration

If enabled, does *not* test the integrity of the IPSI when checking the health of the server pair.

IP Control

Administers IP control of port networks.

Valid Entry	Usage
y	All port networks have an IPSI that provides control. A DS1 Converter (DS1C) circuit pack cannot be added to a port network when this value is y.
n	This IPSI is used only for Tone Clock/Tone Detector functions. Use when the port network contains a DS1 Converter (DS1C) circuit pack.

PRIMARY IPSI

DHCP

Displays whether IPSI is currently set up for DHCP addressing, or static addressing.

- If DHCP is disabled for the System Management Interface, this field is set to disabled (read-only).
- If DHCP is disabled for the IP Server Interface, the following event occurs:
  - If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message.
  - If IPSI is busied out, the static equivalent to the DHCP address automatically populates the **Host**. The **Subnet Mask** and **Gateway** values must be populated manually. The pre-populated values can be optionally overwritten.
- If DHCP is enabled for IP Server Interface, the following event occurs:
  - If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message.
  - If the If the IPSI is busied out, it accepts the changes and re-populates this field.

Gateway

The valid gateway IPv4 address.

- If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access.

- If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

Host

The name of the DHCP client identifier. If DHCP is enabled for the System Management Interface, the **Host** value is displayed for the IP Server Interface. If DHCP is disabled, the **Host** value is not displayed.

IP Address

The valid IPv4 IP address.

- If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access.
- If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

Location

The primary IPSI board location.

Valid Entry	Usage
01 to 64	First and second numbers are the cabinet number
A to E	Third character is the carrier
01 to 20	Fourth and fifth characters are the slot number
01 to 250	Gateway
V1 to V9	Module

Subnet Mask

Valid Entry	Usage
/xx	Represented as subnet bits. • If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access. • If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

QoS AND ETHERNET SETTINGS

802.1p

Valid Entry	Usage
0 to 7	Displays the 802.1p value. This value can be changed only if the System Level Parameter Values is disabled for the IP Server interface. This value takes effect when the IPSI is busied out or released. Accepts only whole numbers. The default value is 6.

Related Links

Use System Level Parameter Values on page 469

Auto

If the IPSI is busied out, you can change this value from disabled to enabled and enabled to disabled.

DiffServ

Valid Entry	Usage
0 to 63	Displays the DiffServ code point (DSCP). This value can be changed only if the System Level Parameters Values is disabled for the IP Server interface. This value takes effect when the IPSI is busied out or released. Default is 46. Accepts only whole numbers.

Related Links

Use System Level Parameter Values on page 469

Duplex

Valid Entry	Usage
Half Full	Displays the duplex settings for this IP board. This value can be changed only if the IPSI is busied out. The default is Full.

Speed

Valid Entry	Usage
10Mbps 100Mbps	Displays the speed of the Ethernet connection. This value can be changed only if the IPSI is busied out. The default value is 100 Mbps.

Use System Level Parameter Values

Enables or disables system level parameter values.

SECONDARY IPSI

DHCP

Displays whether IPSI is currently set up for DHCP addressing, or static addressing.

- If DHCP is disabled for the System Management Interface, this field is set to disabled (read-only).
- If DHCP is disabled for the IP Server Interface, the following event occurs:
  - If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message.
  - If IPSI is busied out, the static equivalent to the DHCP address automatically populates the **Host**. The **Subnet Mask** and **Gateway** values must be populated manually. The pre-populated values can be optionally overwritten.
- If DHCP is enabled for IP Server Interface, the following event occurs:
  - If IPSI is in-service, it disallows the service and displays the `ipserver must be busied out` message.
  - If the If the IPSI is busied out, it accepts the changes and re-populates this field.

Host

The name of the DHCP client identifier. If DHCP is enabled for the System Management Interface, the **Host** value is displayed for the IP Server Interface. If DHCP is disabled, the **Host** value is not displayed.

Gateway

The valid gateway IPv4 address.

- If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access.
- If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

IP Address

The valid IPv4 IP address.

- If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access.
- If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

Location

The secondary IPSI board location.

Valid Entry	Usage
01 to 64	First and second numbers are the cabinet number
A to E	Third character is the carrier
01 to 20	Fourth and fifth characters are the slot number
01 to 250	Gateway
V1 to V9	Module

Subnet Mask

Valid Entry	Usage
/xx	Represented as subnet bits. • If DHCP is enabled for the IP Server Interface, the value of this field is read-only. View the value of this field based on the access. • If DHCP is disabled for the IP Server Interface and this field is changed, after screen validation, the system checks if IPSI is busied out. If not, the IPSI does not accept the change and displays the `ipserver must be busiedout` message.

IP Server Interface (IPSI) Administration: page 2

Dest # 1, 2, or 3 IP Address

The valid destination IPv4 IP address format. Provides support for future IPv6 address format.

Default destination address is unspecified.

Enable Syslog

Enabled only for init and inads login access. For duplicated TN2602 boards, this field is displayed for each board.

Valid Entry	Usage
y	The system checks the firmware support syslog by capabilities exchange information. If it does not get the support, the system displays the `unsupported board code or vintage error` message and does not enable syslog.
n	The system does not display the rest of the syslog fields. This is the default.

Local Facility

Valid Entry	Usage
local0 to local7 (for local use)	Displays the help message on acceptable values. Default value is local4.

Port

Valid Entry	Usage
1 to 65535	The valid port number associated with the **Dest # 1, 2 or 3 IP address** value. Default port number is 514.

Related Links

Dest # 1, 2, or 3 IP Address on page 471

Use System Syslog Values

If enabled, populates the address, port, and facility information as read-only.

DEBUG FILTER VALUES

Object

Valid Entry	Usage
0 to 255	The category of a log event. For example, angel and archangel. **Note:** Maximum value of 255 triggers a file dump.

Level

Valid Entry	Usage
0 to 65535	The level of logging for the given object. **Note:** Maximum value of 65535 triggers a file dump.

Chapter 73: IP Services

Administers the connectivity for various adjuncts.

Note:

You cannot remove a service from this screen if that service has overrides defined on the Survivable Processor screen.

Example command: `change ip-services`

IP Services: page 1

Enabled

Enables or disables this IP service. Available only with an IP **Service Type** of AESVCS or SAT.

Related Links

Service Type on page 474

Local Node

Valid Entry	Usage
IP node name	A previously-administered node name for the link administered for services over the Control Lan (C-LAN) circuit pack.
procr	The Communication Manager's Processor Ethernet interface for adjunct connectivity.

Related Links

Name on page 451

Local Port

The originating port number.

Valid Entry	Usage
5000 to 9999	5111 to 5117 for SAT applications 5678 for ASAI
0	For client applications, defaults to 0.

Remote Node

The server or switch at the far end of the link for SAT.

Valid Entry	Usage
IP node name	A previously-administered node name used to provide added security for SAT.
any	Any available node.

Remote Port

The port number of the destination.

Valid Entry	Usage
5000 to 64500	If this service is a client application, such as CDR or PMS. This value must match the port administered on the adjunct, Personal Computer, or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

The service provided.

Valid Entry	Usage
AESVCS	AE Services.
CDR1, CDR2	Either the primary or secondary CDR device connects over a TCP/IP link.
PMS	Property Management System.
PMS_JOURNAL	The PMS journal printer connects over a TCP/IP link.
PMS_LOG	The PMS log printer connects over a TCP/IP link.
SAT	System administration terminal.
SYS_PRINT	The system printer connects over a TCP/IP link.

IP Services: page 3

Enables reliable protocol for TCP/IP links, and establishes other session-layer parameters. Available only if `CDR1`, `CDR2`, `PMS_JOURNAL`, or `PMS_LOG` is administered for the **Service Type**.

Connectivity Timer

Valid Entry	Usage
1 to 255	The amount of time in seconds that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid Entry	Usage
1 to 255	The number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Enables or disables reliable protocol over this link. If enabled, uses reliable protocol if the adjunct on the far end of the link supports it.

Service Type

The service type for establishing parameters.

Valid Entry	Usage
CDR1, CDR2	Connects either the primary or secondary CDR device over a TCP/ IP link.
PMS_JOURNAL	Connects the PMS journal printer over a TCP/IP link.
PMS_LOG	Connects the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager transmits a unit of protocol data before generating an error.

IP Services: page 4

Creates symbolic name and password pairs for all AE Services servers that are allowed to connect to Communication Manager. Available only if the **Service Type** is AESVCS.

Enabled

Enables or disables the AE Services server.

Password

A password for future access. Accepts 12 to 16 alphanumeric characters and must contain at least one alpha character and one numeric character.

AE Services Server

A valid AE Services Server name. The name must match the AE Services server machine name. Each name must be unique.

Server ID

Valid Entry	Usage
1 to 16	The number assigned to this server.

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Status

Valid Entry	Usage
idle	The AE Services server is connected to Communication Manager.
in-use	The AE Services server is disconnected to Communication Manager.
blank	No AE Server is administered.

Chapter 74: ISDN Trunk Group

Assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group can contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they can also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

Example command: `add trunk-group n`, where *n* is the trunk group number.

ISDN Trunk Group: page 1

Auth Code

The **Auth Code** field is available only for incoming or two-way trunk groups if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk.

The following table provides the situations when the caller must enter an auth code.

Called party	Calling party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the calling station is less than the FRL of the outgoing route pattern
Incoming trunk	Station	No	the **Auth Code** field is enabled on the incoming trunk group

Table continues…

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Called party	Calling party	Facility Restriction Level (FRL) check?	Auth Code, required if
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern or the **Auth Code** field is enabled on the incoming trunk group

Note:

The **Auth Code** field is unavailable if:

- the **Group Type** field is tandem
- the **Group Type** field is ISDN and the **Service Type** field is tandem.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

Related Links

Direction on page 481
Authorization Codes on page 796

Busy Threshold

Valid Entry	Usage
1 to 255	The number of trunks that must be busy to alert attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code are connected to the attendant. Calls handled by AAR and ARS route patterns go out normally.

Carrier Medium

The type of transport medium interface used for the ISDN trunk group.

Valid Entry	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack. Available only if ATM trunking is enabled for the system.
H.323	The trunk is implemented as an H.323 trunk group.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Related Links

Asynch. Transfer Mode (ATM) Trunking on page 795
SBS on page 503

Local Country Code on page 306
International Access Code on page 307
Group Type on page 482
Supplementary Service Protocol on page 495

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this trunk group generate call detail records. If **Record Outgoing Calls Only** is disabled for CDR, incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • **Abandoned calls**: The system creates a record with a condition code of “H” indicating the time until the call was abandoned. • **Answered calls**: The system creates a record with a condition code of “G” indicating the interval from start of ring to answer. • **Calls to busy stations**: The system creates a record with a condition code of “ I ” indicating a recorded interval of 0.

Related Links

Record Outgoing Calls Only on page 106

Charge Advice

Determines how to accumulate and access charge information about a call. Requires that **CDR Reports** be enabled before changing this field from its default of none. Receiving Advice of Charge during the call affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid Entry	Usage
none	The system does not to collect Advice of Charge information for this trunk group.
automatic	The public network sends Advice of Charge information automatically.
end-on-request	Communication Manager requests charge information with each call, and you receive only the final call charge.
during-on-request	Communication Manager requests charge information with each call, and charges display during and at the end of a call.

COR

Valid Entry	Usage
0 to 995	The Class of Restriction (COR) for the trunk group. Classes of restriction control access to trunk groups, including trunk-to-trunk transfers. Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed.

Tip:

Remember that FRLs are assigned to classes of restriction. Even if two trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

Dial Access

Controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

Security alert:

Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and are not restricted by facility restriction levels. For security, leave this field disabled unless dial access is needed to test the trunk group.

Valid Entry	Usage
y	Users can gain access to the trunk group by dialing its access code.
n	Users cannot gain access to the trunk group by dialing its access code. Attendants can still select this trunk group with a **Trunk Group Select** button.

Direction

The direction of the traffic on this trunk group.

Available for all trunk groups except DID and CPE.

Valid Entry	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing
two-way	Traffic on this trunk group is incoming and outgoing. Required for **Network Call Redirection**.

Related Links

Disconnect Supervision-Out on page 493
Answer Supervision Timeout on page 491
Receive Answer Supervision on page 617

Far End Test Line No.

The number sent to the far-end's ISDN test line extension. When the **`test trunk long`** command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. Accepts up to 15 digits.

Example

For an ISDN tandem trunk, the far-end test number should be a seven-digit ETN (Electronic Tandem Network) number.

Group Name

A unique name that provides information about the trunk group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the trunk group rather than the group type (DID, WATS).

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Display Character Set on page 787

Group Number

The trunk group number.

Group Type

The type of trunk group. The fields that are displayed and available might change according to the trunk group type selected.

Valid Entry	Usage
Access	Used to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and, therefore, allow satellite callers unrestricted access to out-dial trunks on the main server. This entry allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks. Used in private networks. This entry allows Inband ANI.
CAMA	Used to route emergency calls to the local community's Enhanced 911 systems.
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
CPE	Used to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Used to direct callers directly to individuals within an organization without going through an attendant or some other central point. This entry allows Inband ANI.
DIOD	Two-way trunks that are used to transmit dialed digits in both directions. In North America, tie trunks are used for applications that require two-way transmission of dialed digits. This entry allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. This entry also allows Inband ANI.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.
ISDN	Used when digital trunks are needed that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Also used for **Network Call Transfer**. **Note:** Available only if **ISDN-PRI**, **ISDN-BRI Trunks**, or both have been enabled for the system. For descriptions of the screens and fields that are unique to ISDN trunks, see *ISDN Trunk Group*.
RLT	Used with Centralized Attendant Service in a private network.
SIP	Used to connect a server running Communication Manager to a Session Manager (SM), or to connect two Communication Manager servers. **Note:** The Automatic CallBack, Priority Calling, and Whisper Page features work incorrectly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Used to connect tandem nodes in a private network. This entry allows Inband ANI.

Table continues…

Valid Entry	Usage
Tie	Used to connect a server running Communication Manager to a local telephone company central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls. This entry also allows Inband ANI.
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can be sent to certain areas (“WATS band”) for a flat monthly charge. Incoming WATS trunks allow toll-free calling to customers and employees.

Related Links

ISDN-PRI on page 801
ISDN-BRI Trunks on page 801
SBS on page 503
Local Country Code on page 306
International Access Code on page 307
Supplementary Service Protocol on page 495
Carrier Medium on page 479
Path Replacement on page 510
Call Still Held on page 880
Call Still Held on page 612

Incoming Calling Number - Format

The TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message excludes CPN information and digits are added, the outgoing message contains these digits. If a numbering format is not administered in this case, the value defaults to pub-unk. If the numbering format is administered as unknown, the trunk group is modified to unk-unk encoding of the TON/NPI. Therefore, this field also must contain a value other than unknown.

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid Entry	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)

Table continues…

Valid Entry	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Related Links

Format on page 885
Format on page 499

Incoming Calling Number Insert

Valid Entry	Usage
0 to 9 all blank	The number of digits inserted in the calling party number for all incoming calls on this trunk group. Accepts up to 15 characters.

Member Assignment Method

Available only if the **Carrier Medium** is H.323.

Valid Entry	Usage
manual	Users manually assign trunk members to a signaling group. This is the default.
auto	The system automatically generates members to a specific signaling group.

Related Links

Carrier Medium on page 479

Number of Members

Available only if the **Carrier Medium** is H.323 and the **Member Assignment Method** is auto.

Valid Entry	Usage
0 to 255	The number of virtual trunk members automatically assigned to the signaling group. Default is 0.

Related Links

Carrier Medium on page 479

Member Assignment Method on page 485
Signaling Group on page 488

Outgoing Display

With Outgoing Display, telephones can show the name and number of the trunk group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Queue Length

Available only for outgoing or two-way trunk groups.

Valid Entry	Usage
1 to 100	The number of outgoing calls that can wait in queue when all trunks in a trunk group are busy. Calls wait in queue in the order in which they were made. If a queue is administered, a caller hears a confirmation tone when no trunk is available for the outgoing call. The caller can then disconnect and wait; when a trunk becomes available, Communication Manager calls the extension that placed the original call. Communication Manager remembers the number the caller dialed and automatically completes the call.
0	Callers receive a busy signal when no trunks are available. Use for DCS trunks.

Related Links

Direction on page 481

Service Type

The service for which this trunk group is dedicated. In addition to the predefined services or features listed as valid entries, any previously administered user-defined Network Facility **Facility Type** of 0 (feature) or 1 (service) is allowed.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.

Table continues…

Valid Entry	Usage
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — A subscriber can receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, **Dial Access** can be enabled.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related Links

Facility Type on page 600

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Signaling Group

Available only if the **Carrier Medium** is H.323 and the **Member Assignment Method** is auto.

Valid Entry	Usage
digits or blank	Assigned H.323 signaling group number. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Related Links

Carrier Medium on page 479
Member Assignment Method on page 485

TAC

The trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group. The characters “*” and “# ” can be used as the first character in a TAC. Accepts a one- to four-digit number.

TestCall BCC

The Bearer Capability Code (BCC) used for the ISDN test call.

Valid Entry	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability Information Element (IE) in the `SETUP` message when generating an ISDN test call.

Note:

The ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Valid Entry	Usage
rest	Restricted
unre	Unrestricted

Testcall Service

The call-by-call selection for an ISDN test call. Available only if the **Service Type** is cbc. Unavailable for **Facility Type** 0 (feature), 1 (service), or 3 (outgoing) that is defined by users.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, **Dial Access** can be enabled.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.

Table continues…

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Valid Entry	Usage
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related Links

Service Type on page 486
Facility Type on page 600

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Usage Alloc

If enabled, allocates service provided by the trunk group and enhances Network Call Redirection. Available only if the ISDN Trunk Group **Service Type** is cbc.

Related Links

Service Type on page 486

ISDN Trunk Group: page 2

Administer Timers

Enables or disables administration of timers on this trunk group. The default for the ISDN trunk group type is disabled. All other trunk group types are enabled by default.

Available for all trunk group types *except* cpe, h.323, and sip.

Related Links

Group Type on page 482

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Note:

This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers.

Related Links

Receive Answer Supervision on page 617
Administer Timers on page 490

Bit Rate

Valid Entry	Usage
300 1200 2400 4600 9600 19200	The baud rate used by pooled modems. The speed of the fastest modem that uses this ISDN trunk group.

Charge Advice

Determines how to accumulate and access charge information about a call. Requires that **CDR Reports** be enabled before changing this field from its default of none. Receiving Advice of Charge during the call affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid Entry	Usage
none	The system does not to collect Advice of Charge information for this trunk group.
automatic	The public network sends Advice of Charge information automatically.

Table continues…

Valid Entry	Usage
end-on-request	Communication Manager requests charge information with each call, and you receive only the final call charge.
during-on-request	Communication Manager requests charge information with each call, and charges display during and at the end of a call.

Codeset to Send Display

Defines the codeset for sending the information element for display. The value depends on the type of server or switch used for the connection.

Valid Entry	Usage
0	CCITT (non-Communication Manager equipment).
6	Any other than CCITT or System 85 R2V4, 4E11.
7	System 85 R2V4, 4E11.

Codeset to Send National IEs

The codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of server/switch to which the user is connected.

Valid Entry	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer Avaya S8XXX Server types.

Note:

A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the **Service Type** is tandem. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

CONNECT Reliable When Call Leaves ISDN

Available only if **Group Type** is ISDN. The value tells the Communication Manager server whether a CONNECT received on an outgoing call that is not end-to-end ISDN is a reliable indication that the far end has answered the call.

Valid entry	Usage
y	The CONNECT is a reliable answer.
n	The CONNECT message is an unreliable answer. The message can be the result of a timer expiring. If the call was originated by a Call Center adjunct, a Call Classifier can be used instead to determine whether the call has been answered. The default value is n.

Digit Handling (in/out)

Valid Entry	Usage
enbloc/enbloc enbloc/overlap overlap/enbloc overlap/overlap	Defines whether overlap receiving and overlap sending features are enabled. enbloc disables overlap receiving and overlap sending. The first field value indicates digit receiving and the second value indicates digit sending.

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group. If values other than 18 or between 11 and 15 are administered, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Disconnect Supervision-Out

Indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. Available for outgoing or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end sends a release signal when the called party releases an outgoing call, and the far-end is responsible for releasing the trunk. Enhances Network Call Redirection. Available only if **Answer Supervision Timeout** is 0 and **Receive Answer Supervision** is enabled.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on outgoing calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

Verify that the far-end server or switch provides answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with the network services provider.

Related Links

Direction on page 481
Answer Supervision Timeout on page 491
Receive Answer Supervision on page 617

Duplex

Note:

Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid Entry	Usage
full	Provides simultaneous two-way transmission, which is most efficient. Recommended in most cases.
half	Supports only one transmission direction at a time.

Group Type

Displays the type of trunk group.

Related Links

Group Type on page 482
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Max Message Size to Send

Valid Entry	Usage
128 244 256 260	The maximum number of bytes of an ISDN message for Communication Manager.

The following table indicates the expected ISDN-PRI message size from several Lucent Technologies and Avaya Inc. products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

Supplementary Service Protocol

The supplementary service protocol to use for services over this trunk group. Supplementary service protocols are mutually exclusive.

Valid Entry	Usage
a	National.
b	ISO/ETSI QSIG Private Network. Also used for SBS signaling trunks.
c	ETSI public network.
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC)).
e	DCS with Rerouting. Do not use with **Service Type** of dmi-mos or sddn.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI.

Related Links

SBS on page 503
Local Country Code on page 306
International Access Code on page 307
Group Type on page 482
Carrier Medium on page 479

Synchronization

Caution:

Do not change this field without the assistance of Avaya or the network service provider.

Valid Entry	Usage
async sync	Determines whether the trunk group uses synchronous or asynchronous communications.

Trunk Hunt

Defines the trunk hunt search order. Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Valid Entry	Usage
ascend	Enables a linear trunk hunt search from the lowest to highest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.
cyclical	Enables a circular trunk hunt based on the sequence the trunks were administered within the trunk group. When using ISDN-BRI interfaces, only cyclical is supported. The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations.
descend	Enables a linear trunk hunt search from the highest to lowest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.

Related Links

Wideband Support on page 508

Trunk Type

The type of trunk.

Related Links

Trunk Type (in/out) on page 876

ISDN Trunk Group: page 3

ACA Assignment

Indicates whether Automatic Circuit Assurance (ACA) measurements are taken for this trunk group.

Apply Local Ringback

Enables or disables local ringback tone to the caller. If enabled, local ringback is removed when the call is connected. Available only if the **Carrier Medium** is PRI_BRI.

Related Links

Carrier Medium on page 479

BSR Reply-best DISC Cause Value

Servers running Communication Manager that are polled as resources in a Best Service Routing application return data to the polling server in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that the server returns in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group are dropped before any data is returned to the polling server or switch.

Available only if **UUI IE Treatment** is shared.

Valid Entry	Usage
31	Normal-unspecified. This value is almost always used.
17	User-busy
16	Normal-call-clearing.

Caution:

In most cases, this field is set to the appropriate value during installation. Do not change this field without the assistance of Avaya or your network service provider.

Related Links

UUI IE Treatment on page 508

Charge Conversion

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related Links

Direction on page 481
Trunk Direction on page 617

Currency Symbol

The telephone displays the symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for ISDN trunk groups, if the Charge Advice field is set to a value other than none.

Related Links

Direction on page 481
Trunk Direction on page 617

Data Restriction

If y, whisper page is denied on this trunk. Enter y to prevent features from generating tones on a data call that would cause erroneous data transmission.

Decimal Point

The appropriate representation for a decimal point as telephone displays it. Available only with outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field is independent whether the calling display shows a decimal point. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related Links

Direction on page 481
Charge Advice on page 480

DS1 Echo Cancellation

Enables or disables echo cancellation on a per port basis. If enabled, reduces voice call echo.

Note:

Changes to the DS1 Echo Cancellation field is incomplete until one of the following occurs:

- Port is busied-out or released.
- Trunk group is busied-out or released.
- SAT command test trunk group is performed.
- Periodic maintenance runs.

DSN Term

Enables or disables the trunk group as a DSN termination telephone. The default is disabled.

Format

Specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. If you have used Special Application SA9118 to generate international numbers in a Service Pack for Communication Manager Releases 5.2.1 or 6.0.1, you must set the **Format** field to intl-pub for the appropriate ISDN trunk group(s) after upgrading a Communication Manager server to Release 6.2 or later, because there is no upgrade path for intl-pub entry in Communication Manager 6.2.

The **Format** field is available only if the **Send Calling Number** or **Send Connected Number** field is enabled or restricted.

Note:

The entries private and unk-pvt are overridden, and a public number is generated, if the called number for the outgoing call over the trunk group is encoded using E.164. The numbering plan for the called number is determined by the **Call Type** field from the ARS or AAR Analysis, or the **Numbering Format** field from the outgoing Route Pattern.

Valid Entry	Usage
intl-pub	Indicates the **Numbering Plan Indicator** is E.164. International (public number that includes city or area code and the country code).
natl-pub	Indicates the **Numbering Plan Indicator** is E.164. National (public number that includes city or area code, but excludes the country code).
private	Indicates the **Numbering Plan Indicator** is Private. Based on the network level of the **ISDN Parameters** field on the Feature-related System Parameters screen.
pub-unk	Indicates the **Numbering Plan Indicator** is E.164. Unknown (public number with no details of the digits included).
unk-pvt	Indicates the **Numbering Plan Indicator** is Private. Unknown (private number with no details about the digits included).

Related Links

Send Calling Number on page 504
Send Connected Number on page 505
Numbering-Private Format on page 602

Maintenance Tests

Enables or disables hourly maintenance tests on this trunk group.

Available only for aplt, isdn, sip, or tie trunk groups.

Related Links

Group Type on page 482

Maximum Size of UUI IE Contents

Available only if the **UUI IE Treatment** is shared.

Valid Entry	Usage
32 to 128	The maximum number of bytes of user information that the network supports.

Related Links

UUI IE Treatment on page 508

Measured

Indicates if the system transmits data for this trunk group to the Call Management System.

Valid Entry	Usage
internal	Sends the data to the Basic Call Management System (BCMS), the VuStats data display, or both. Available only if **BCMS (Basic)** or **VuStats** is enabled for the system.
external	Sends the data to the CMS.
both	Collects data internally and sends it to the CMS. Available only if **BCMS (Basic)** or **VuStats** is enabled for the system.
none	Trunk group measurement reports are not required.

Related Links

VuStats on page 812
BCMS (Basic) on page 807

Modify Tandem Calling Number

Available with outgoing or two-way trunks when the **Carrier Medium** field is set to PRI/BRI or H.323, and the **Send Calling Number** field is set to enabled or restricted.

Valid Entry	Usage
natl-intl-prefix	Adds the national or international prefixes from the Locations Parameters screen when the calling party number is appropriate.
tandem-cpn-form	Modifies the calling party number IE in the previously administered format specified for the Tandem Calling Party Number.
no	Does not modify the calling party number.

Related Links

Group Type on page 482
Direction on page 481
Carrier Medium on page 479
Send Calling Number on page 504
Modify Tandem Calling Number on page 901

NCA-TSC Trunk Member

The trunk member number whose D-channel is used to route tandem NCA-TSCs or QSIG CISCs.

Network Call Redirection

Whenever the **Supplementary Service Protocol** is changed, this field resets to none to prevent an inadvertent incorrect value.

Available only if **ISDN Network Call Redirection** is enabled for the system and the **Supplementary Service Protocol** is a, c, or g.

Related Links

ISDN-PRI on page 801
ISDN/SIP Network Call Redirection on page 802
Supplementary Service Protocol on page 495

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the disconnect message.

Outgoing Channel ID Encoding

Determines whether to encode the Channel ID IE as preferred or exclusive. Available only if the **Group Type** is isdn and the **Service Type** is anything except dmi-mos or sddn. Blank is an invalid entry.

Valid Entry	Usage
preferred	Default if **Used for DCS** is disabled or unavailable.
exclusive	Default if **Used for DCS** is enabled.

Related Links

Group Type on page 482
Service Type on page 486

Path Replacement Method

Available only if **Basic Call Setup** and **Supplementary Services with Rerouting** are enabled for the system and when the **Supplementary Service Protocol** is either b or e and the **Group Type** is isdn.

Valid Entry	Usage
better-route	Uses the most economical route; for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Related Links

ISDN-PRI on page 801
ISDN-BRI Trunks on page 801
Basic Call Setup on page 813
Supplementary Services with Rerouting on page 814
Service Type on page 486
Group Type on page 482

Replace Restricted Numbers

Indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. If enabled, the display is replaced regardless of the service type of the trunk. Applies to BRI, PRI, H.323, and SIP trunks. Available only if the **Group Type** is isdn or sip.

Related Links

Group Type on page 482

Replace Unavailable Numbers

If enabled, replaces unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. The display is replaced regardless of the service type of the trunk. Applies to BRI/PRI, H.323, and SIP trunks. Also applies to analog trunks if **Analog Trunk Incoming Call ID** is enabled and **Receive Analog Incoming Call ID** is set to any value except disabled. Available only if the **Group Type** is isdn or sip.

Related Links

Group Type on page 482
Analog Trunk Incoming Call ID on page 794
Receive Analog Incoming Call ID on page 904

SBS

Enables or disables Separation of Bearer and Signaling (SBS) for the trunk group.

Available only if the **Local Country Code** and **International Access Code** are administered for the system and when the **Supplementary Service Protocol** is b, the **Group Type** is isdn, the **Carrier Medium** is H.323, and **Dial Access** is disabled.

Related Links

Local Country Code on page 306
International Access Code on page 307
Group Type on page 482

Supplementary Service Protocol on page 495
Carrier Medium on page 479

Send Called/Busy/Connected Number

Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is enabled for the trunk group.

Valid Entry	Usage
y	The dialed number is sent on incoming or tandemed ISDN calls. This field must be enabled in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the dialed number on incoming or tandemed ISDN calls.
r	Restricted. The connected number is sent “presentation restricted”.

Related Links

Numbering — Public/Unknown Format on page 604
QSIG Value-Added on page 511
Numbering-Private Format on page 602

Send Calling Number

Specifies whether the calling party’s number is sent on outgoing or tandemed ISDN calls.

Note:

The Numbering - Public/Unknown Format can override the Send Calling Number administration

Valid Entry	Usage
y	The calling party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the calling party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The calling number is sent “presentation restricted”. If set to restricted, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Related Links

Numbering-Private Format on page 602

Incoming Number Format on page 96
Numbering — Public/Unknown Format on page 604

Send Codeset 6/7 LAI IE

If enabled, the ISDN trunk transmits information in Codeset 6/7.

If the **UUI IE Treatment** is shared, then this field should be disabled. Otherwise, the same information will be sent twice and might exceed the message size.

Related Links

UUI IE Treatment on page 508

Send Connected Number

Specifies if the connected party's number is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is disabled for the trunk group.

Valid Entry	Usage
y	The connected party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used. This field must be enabled for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.
n	Disables the sending of the connected party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The connected number is sent "presentation restricted". If this field is set to r, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination for the DS1 Circuit Pack, you should not administer the Send Connected Number as restricted, as this causes display problems. The Numbering - Public/Unknown Format overrides the Send Connected Number administration for any administrable block of extensions.

Related Links

Numbering-Private Format on page 602
QSIG Value-Added on page 511
Numbering — Public/Unknown Format on page 604

Send Name

Specifies whether the calling/connected/called/busy party's administered name, or the name on a redirected call, is sent to the network on outgoing/incoming calls.

Note:

If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Valid Entry	Usage
y	The calling/connected/called/busy party's administered name and the name on a redirected call are sent on outgoing/incoming calls. An entry of y or n is required if the **Supplementary Service Protocol** is e (DCS with Rerouting).
n	Disables the sending of the calling/connected/called/busy party's administered name and the name on a redirected call on outgoing/incoming calls. An entry of y or n is required if the **Supplementary Service Protocol** is e (DCS with Rerouting). With an entry of n, an incoming name is not tandemed out again if the **Supplementary Service Protocol** field is any value other than b (QSIG).
r	Restricted. The calling/connected/called/busy party's administered name is sent "presentation restricted" on outgoing/incoming calls. The name on a redirected call is not sent. This value is valid only if the **Supplementary Service Protocol** is a (national supplementary service), b (for called/ busy only), or d (for the QSIG Global Networking Supplementary Service Protocol). With an entry of r, an incoming name is marked restricted when it is tandemed out again. However, if the **Supplementary Service Protocol** is b (QSIG), then an incoming name is passed on unchanged and **Send Name** value is ignored.

Related Links

Supplementary Service Protocol on page 495

Send UCID

If you enable the **Send UCID** field, the trunk transmits Universal Call IDs (UCIDs). This field is available only if the **UUU IE Treatment** field is set to `Shared`.

Note:

If the incoming trunk has two UCIDs in the UUI, the outgoing trunk sends the UUI with both UCIDs to the far-end switch.

Send UUI IE

If enabled, sends UUI information on a per trunk group basis.

Show ANSWERED BY on Display

Available only for isdn pri/bri, isdn h.323, and sip trunk groups.

Valid Entry	Usage
y	The words “ANSWERED BY” display in addition to the connected telephone number on calls over this trunk. This is the default. **Note:** Based on display language settings for stations, “ANSWERED BY” is translated into and displayed in the appropriate language.
n	Only the connected telephone number displays. This might be preferred when outgoing calls are over a trunk that might be redirected.

Suppress # Outpulsing

Indicates whether or not to suppress the final “#” in cases where the system would normally outpulse it. Used if end-to-end signaling begins with (and includes) “#”. This field should be enabled when the local telephone company central office or any other facility treats “#” as an error.

US NI Delayed Calling Name Update

For digital display stations, if calling name information is received after the incoming call has been delivered to the terminating telephone, enables or disables a display update.

For analog display stations, Communication Manager is enhanced to wait for the FACILITY message (that contains the name information) before delivering an incoming call to the receiver. You can administer the time duration for which Communication Manager waits for the FACILITY message.

Note:

BRI trunks do not support display updates.

Available only if the **Carrier Medium** is either PRI/ BRI or ATM, and the **Supplementary Service Protocol** is a.

Related Links

Supplementary Service Protocol on page 495
Carrier Medium on page 479
ISDN-PRI on page 801

UUI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid Entry	Usage
shared	The trunk is connected to the server.
service-provider	Service provider functionality is required.

Wideband Support

Note:

This feature is not supported on the DS1 interfaces on branch gateways (G700/G350).

Enables or disables wideband switching on this trunk group. Only trunk members from TN464C or later circuit packs can use wideband switching.

Available only if **Wideband Switching** is enabled for the system.

Note:

Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Related Links

Wideband Switching on page 807

QSIG Trunk Group Options

The fields on this screen appear only when **Group Type** is isdn and **Supplementary Service Protocol** is b.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 495
Trunk Group on page 860

Character Set for QSIG Name

Use this field to configure the character set for transmission and display of the QSIG name data. This field is available only if:

- **Group Type** is isdn.
- **Supplementary Service Protocol** is b.
- **Display Character Set** is Roman.

Valid Entry	Usage
eurofont	The Roman Eurofont character set. Eurofont is the default setting. **Note:** Avaya BRI stations support only ASCII characters. A BRI station does not correctly display non-ASCII characters, such as Eurofont or Kanafont.
iso-8859-1	The system converts all data or characters transmitted over a QSIG trunk from Eurofont to ISO 8859-1. Eurofont is an Avaya proprietary encoding character set. **Note:** ISO 8859-1, also known as ISO/IEC 8859-1 or Latin-1, is part 1 of ISO/IEC 8859, a standard character encoding defined by ISO. ISO 8859-1 encodes Latin alphabet no. 1, which consists of 191 characters from the Latin script. Each character is encoded as a single 8-bit code value.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 495
Display Character Set on page 787

Diversion by Reroute

Use this field to activate or deactivate the Diversion by Reroute feature. The default value of this field is y.

This field is available only if:

- **Group Type** is ISDN.
- **Supplementary Service Protocol** is b.

If the value of this field is y , Communication Manager selects the optimum path and routes the call on that path to the destination number.

If the value of this field is n, Communication Manager does not initiate a Diversion or Reroute request over the QSIG trunk. Communication Manager also rejects any Diversion or Reroute request that Communication Manager receives over the QSIG trunk.

Related Links

Supplementary Service Protocol on page 495
Group Type on page 482

Path Replacement

Enables or disables the Path Replacement feature. The default is enabled. If disabled, Communication Manager does not originate a Path Replacement request over that trunk group, and rejects any Path Replacement request it receives over that trunk group. Available only if the **Group Type** is isdn and the **Supplementary Service Protocol** is b.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 495

Path Replacement Method

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system. Unavailable if **Path Replacement with Retention** is enabled.

Valid Entry	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if **Path Replacement with Retention** is enabled. Path replacement fails (and the original trunk group is retained) if **Path Replacement with Retention** is disabled.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 495
Supplementary Services with Rerouting on page 814
DCS with Rerouting on page 797
Path Replacement with Retention on page 510

Path Replacement with Retention

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system.

Valid Entry	Usage
y	Retains the original trunk group.
n	Provides path replacement according to settings for the **Path Replacement Method** .

Related Links

Supplementary Service Protocol on page 495
Supplementary Services with Rerouting on page 814
DCS with Rerouting on page 797
Group Type on page 482

QSIG Value-Added

Enables or disables QSIG-VALU services. Available only if **Value-Added (VALU)** is enabled for the system and **Supplementary Services Protocol** is enabled for the trunk.

Related Links

Value Added (VALU) on page 815
Supplementary Service Protocol on page 495

QSIG-Value Coverage Encoding

The encoding method used to encode DL1, DL2, and DL3 extensions. Available only if the **Group Type** is isdn, **Supplementary Service Protocol** is b, and **QSIG Value-Added** is enabled for the system.

Valid Entry	Usage
proprietary	Communication Manager sends extension information in the normal manner. This is the default.
standard	In addition to normal extension information, Communication Manager sends the data part (as null) of the extension.

Related Links

Value Added (VALU) on page 815
Group Type on page 482
Supplementary Service Protocol on page 495

SBS

Enables or disables Separation of Bearer and Signaling (SBS) for the trunk group.

Available only if the **Local Country Code** and **International Access Code** are administered for the system and when the **Supplementary Service Protocol** is b, the **Group Type** is isdn, the **Carrier Medium** is H.323, and **Dial Access** is disabled.

Related Links

Local Country Code on page 306
International Access Code on page 307
Group Type on page 482
Supplementary Service Protocol on page 495
Carrier Medium on page 479

SIP Reference Trunk Group

The system displays SIP Reference Trunk Group only when the **Group Type** field is isdn, the **Carrier Medium** field is H.323, and the **Supplementary Service Protocol** field is b.

Valid Entry	Usage
1 to 2000	Assigns a number for the SIP trunk group. If trunk members are already assigned to this trunk group, you must not change the value of **SIP Reference trunk Group** field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
blank	No SIP trunk group is assigned. By default, the value is blank.

Note:

For the maximum allowed entry per server type, see Avaya Aura® Communication Manager System Capacities Table, 03–300511.

TSC Method for Auto Callback

Controls the signaling connection method for the QSIG Temporary Signaling Connections (TSC) when Communication Manager is the terminating or the outgoing gateway PINX.

Valid Entry	Usage
drop-if-possible	QSIG Temporary Signaling Connections are released.
always-retain	QSIG Temporary Signaling Connections are always retained for successful call completion activation.

ISDN Trunk Group: Administrable Timers

This screen displays only when **Administer Timers** is enabled. This screen does not display ISDN trunk group information for trunks of **Group Type** cpe or sip.

Note:

If the ISDN trunk group has a **Carrier Medium** value of H.323, or if the trunk group has BRI members, then this page cannot be administered. In these cases, an error message displays when you attempt to submit the screen.

Caution:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Related Links

Group Type on page 482
Administer Timers on page 490
Carrier Medium on page 479

Programmed Dial Pause (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The exact duration of the pause used during abbreviated dialing, ARS outpulsing, and terminal dialing operations. This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All central office (CO) circuit packs that accept administrable timers accept this timer.

END TO END SIGNALING

Pause (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The interval (pause) between DTMF tones sent from a hybrid telephone. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Related Links

Trunk Type on page 894

Tone (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the DTMF tone sent when a button on a hybrid telephone is pressed. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Related Links

Trunk Type on page 894

Shared UUI Feature Priorities

The fields in this page show the priorities for each type of information to be forwarded in the Shared UUI. The system displays this page only on the ISDN trunk group screen when all of the following conditions are met:

- The **UUI IE Treatment** is shared.
- The **Supplementary Service Protocol** is set to anything except b.

Changing the priorities in this screen might affect whether certain information will be sent.

Related Links

Supplementary Service Protocol on page 495
UUI IE Treatment on page 508

ASAI

User information from Adjunct/Switch Applications Interface (ASAI).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 1.
blank	This field's information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 5.
blank	This field's information is not forwarded.

Held Call UCID

The unique tag for the last call that was put on hold by the Automatic Call Distribution (ACD) agent placing this call to another system. This Universal Call ID (UCID) can be used to identify the original or parent call that may eventually be placed into conference or transferred to the other system. This element is required for cradle-to-grave tracking with Avaya IQ release 5.0 and later.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. The UCID included in the element with default priority 2 is the tag for a new call placed by the agent while the original call is on hold. Default priority is 7.
blank	This field's information is not forwarded.

In-VDN Time

Number of seconds the call has spent in vector processing.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 3.
blank	This field's information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call’s priority level in its current queue, and the type of interflow.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 6.
blank	This field's information is not forwarded.

Universal Call ID

A unique tag that identifies the call that this message is being sent for and the other information included in the User-User-Information (UUI).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 2.
blank	This field's information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 4.
blank	This field's information is not forwarded.

CBC Service Trunk Group Allocation Plan Assignment Schedule

This screen administers a fixed schedule or a schedule that can change up to six times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan will be in use at any given time.

Available only if the ISDN trunk group **Service Type** is cbc and the ISDN trunk group **Usage Alloc** are enabled.

Related Links

Service Type on page 486

Usage Alloc on page 490

Act Time

The time the usage allocation plan will become effective.

Valid Entry	Usage
00:00 through 23:59	The time using the 24–hour clock (military time). There must be at least one entry per day.

Allocation Plan Number

Valid Entry	Usage
1 to 3	The CBC Trunk Allocation Plan that is in effect if a fixed usage method has been selected. Required if the allocation plan is fixed.

Valid Entry	Usage
blank	

Fixed

Indicates whether the allocation plan will be fixed.

Plan

Valid Entry	Usage
1 to 3 blank	The number of the usage allocation plan that will be in effect from the activation time until the activation time of the next scheduled plan change.

Scheduled

Indicates whether or not the allocation plans are implemented according to the administered schedule found on this page. If enabled, then there must be at least one entry in the schedule.

CBC Trunk Group Usage Allocation

This screen sets a minimum and maximum number of members for up to ten different Services/ Features for up to three different Usage Allocation Plans (1 to 3). Available only if the **Service Type** is cbc and **Usage Alloc** is enabled.

Related Links

Service Type on page 486
Usage Alloc on page 490

Max# Chan

Valid Entry	Usage
0 to 99 blank	The maximum number of members of an ISDN trunk group with a **Service Type** of cbc that a particular service or feature can use at any given time. This field must be administered if a service or feature has been entered in the Incoming Call Handling Treatment Table screen.

Min# Chan

Valid Entry	Usage
0 to 99 blank	The minimum number of members of an ISDN trunk group with a **Service Type** of cbc that a particular service or feature can use at any given time. The sum of the minimum number of members for all services or features must not exceed the total number of members of the trunk group.

Service/Feature

The ISDN services or features that can be requested at call setup time when using this trunk group. In addition to predefined services or features that can be received on a call by call basis, user-defined service types can be used.

Valid Entry	Usage
0	Feature
1	Service
2	Incoming
3	Outgoing
other	Any services or features that are not explicitly specified

Wideband Support Options

Note:

All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility. Also, all trunk members in an ISDN trunk group with **Wideband Support** enabled must be from a TN464C (or later) circuit pack.

Contiguous

Available only if “N by DS-zero” (NXDS0) multi-rate service is enabled.

Valid Entry	Usage
y	Specifies the “floating” scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk). Unavailable with H0 ISDN information transfer rate.

Table continues…

Valid Entry	Usage
n	Specifies the “flexible” scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points

Related Links

NxDS0 on page 519
NXDS0 on page 633

H0

If enabled, specifies the ISDN information transfer rate for 384-kbps of data that is comprised of six B-channels. When a trunk group is administered to support H0, the trunk or hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

If enabled, specifies the ISDN information transfer rate for 1536-kbps of data that is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk or hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

If enabled, specifies the ISDN information transfer rate for 1920-kbps of data that is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk or /hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

NxDS0

Enables or disables the "N by DS-zero" multi-rate service.

ISDN Trunk Group: Group Member Assignments

Note:

When supporting DCS, Member Number Assignments must be the same between nodes. For example, Member #1 must be Member #1 at the far-end trunk group.

Administered Members (min/max)

The minimum and maximum member numbers that have been administered for this trunk group.

Code

The type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address, the field is blank.

Name

The name of the trunk group member. The name should identify the trunk unambiguously. Accepts up to 10 characters.

Example

- The telephone number assigned to incoming trunks
- The Trunk Circuit Identification number assigned by the service provider

Night

The night service destination for this trunk group member if different from the night service destination administered for the trunk group. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid Entry	Usage
A valid extension	The extension of the night destination for the trunk.
attd	Calls go to the attendant when night service is active.
blank	Not administered

Related Links

Night Service on page 870

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1’s port number is 01A1002, then B channel 2’s port number is 01A1018.

When using ISDN-PRI interfaces, the port number will be the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

Note:

When administering analog trunks connected to a TIM518, physical ports 17 to 24 are administered as ports 9 to 16 in Communication Manager.

Sfx

The model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address, the field is blank.

Sig Grp

Valid Entry	Usage
digits	The signaling group of this trunk group member. Required for IP group members. If you administer a port that resides on a DS1 board and that DS1 board belongs to one and only one signaling group, you can leave the **Signaling Group** blank. The appropriate default signaling group number is inserted by Communication Manager. If a DS1 board is assigned to more than one signaling group, then you must enter a signaling group number. A trunk group can contain members from different signaling groups. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Related Links

Group Type on page 482
Signaling Group on page 488

Total Administered Members

The total number of members administered in the trunk group.

Chapter 75: ISDN-BRI Trunk Circuit Pack

Administers an ISDN-BRI circuit pack.

Example command: `change bri-trunk-board n`, where *n* is the board location.

ISDN-BRI Trunk Circuit Pack: page 1, TN2185 circuit pack

Cntry/Peer Protocol

The ISDN protocol standard that is applied.

Valid Entry	Usage
1 to 25	An entry of 10, 12, or 13 is equivalent to a **Protocol Version** of b administered for the DS1 Circuit Pack. All other administered values are equivalent to a **Protocol Version** of a administered for the DS1 Circuit Pack.
etsi	Equivalent to a **Protocol Version** of b administered for the DS1 Circuit Pack.
QSIG	Required for a peer-slave or peer-master interface. Valid only when the Interface field is peer-slave.
blank	Cannot be blank if an interface is administered.

Related Links

Protocol Version on page 218
Interface on page 523

DCP/Analog Bearer Capability

Valid Entry	Usage
3.1kHz speech	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.

Detect Slips

The system displays the field only if you set the **Termination Type** field to TE. Use the field to determine the state of health due to slips. This field is applicable only to the TN2185 boards.

ETSI CCBS

Available only if **TSC SS Protocol** is c.

Valid Entry	Usage
none	Interface supports neither incoming nor outgoing ETSI CCBS. This is the default.
inco(ming)	Interface supports only incoming ETSI CCBS.
outg(oing)	Interface supports only outgoing ETSI CCBS.
both directions	Interface supports incoming and outgoing ETSI CCBS. **Note:** When upgrading from a version of Communication Manager that is earlier than 5.1, this is the default.

Related Links

Group Type on page 482

Interface

Valid Entry	Usage
network user peer-master peer-slave	Indicates whether a particular port is connected to a user, network or a peer interface. These entries are valid for the TN2185. Peer-slave is available only if the QSIG Basic Call Setup feature is enabled.

Interface Companding

The companding algorithm expected by the system at the far end.

Valid Entry	Usage
a-law	Algorithm expected at the far-end for E1 service.
mu-law	Algorithm expected at the far-end for T1 service.

Layer 1 Stable

The system displays the field only if you set the **Termination Type** field to TE.

Valid Entry	Usage
y	The far-end network is stable at Layer 1.
n	The far-end network can drop Layer 1 after a call is completed and near-end ignores the Layer 1 disconnect message.

Location

Displays the TN2185 circuit pack location (PPCSS).

Name

The name used to identify the circuit pack. Accepts up to 15 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Port

The port number to which administered parameters apply.

Side

Valid Entry	Usage
a b	Determines how glare conditions are handled for a peer-slave interface.

Related Links

Interface on page 214

Synch Source

Valid Entry	Usage
y	You can administer the board in the change synchronization screen as a primary or secondary Synch Source.
n	You cannot administer the board in the change synchronization screen as a primary or secondary Synch Source.

Note:

The Synch Source field, when displayed, applies to TN2185, MM720, and MM721 boards.

T3 Timer Length (sec)

Valid Entry	Usage
1 to 127	Tells the TE side how many seconds to wait for an inactive Layer 1 to become active.

TEI

Valid Entry	Usage
auto	TEI is assigned automatically by the network.
0	TEI is fixed.

Termination Type

When a MM720 media module is used as a trunk interface, and the MM720 supports both Line side and Trunk side of BRI, indicates whether the media module is to operate in Terminal or Network termination mode.

When you change the **Termination Type** of the MM720/MM721 media module from TE to NT, the system checks if the media module is administered as a clock synchronization source in the Synchronization Source screen, which is the circuit packet location. After the system completes the checks, the system displays the following error message if the board is administered as a synchronization source: `This board is used for synchronization; cannot change Termination Type.`

If you change the **Termination Type** of the MM720/MM721 media module from NT to TE and set any of the interface fields to network or peer-master for a port, the system displays the following warning message: `TE Termination Type not recommended with network Interface types.`

Likewise, if you change the **Termination Type** of the MM720/MM721 media module from TE to NT and set any of the interface fields to user or peer-slave for a port, the system displays the following warning message: `NT Termination Type not recommended with network Interface types.`

However, you can still submit these changes, if required.

Valid Entry	Usage
TE	Terminal Endpoint termination. The MM720 provides the TE side of the BRI interface. This is default.
NT	Network Termination. The MM720 provides the NT side of the BRI interface. The NT type board cannot be used as a clock synchronization source.

ISDN-BRI Trunk Circuit Pack: page 1, TN556B or TN2198 circuit packs

The following field descriptions are unique to the ISDN-BRI Circuit Pack screen with a TN556B or TN2198 circuit pack. The following fields do not display with a TN556B or TN2198 circuit pack:

- **T3 Timer Length (sec)**
- **Synch Source**
- **Layer 1 Stable**
- **Detect Slips**

DCP/Analog Bearer Capability

Valid Entry	Usage
3.1kHz speech	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.

Cntry/Peer Protocol

The ISDN protocol standard that is applied.

Valid Entry	Usage
1 to 25	An entry of 10, 12, or 13 is equivalent to a **Protocol Version** of b administered for the DS1 Circuit Pack. All other administered values are equivalent to a **Protocol Version** of a administered for the DS1 Circuit Pack.

Table continues…

Valid Entry	Usage
etsi	Equivalent to a **Protocol Version** of b administered for the DS1 Circuit Pack.
QSIG	Required for a peer-master interface.
blank	This field cannot be blank if an interface is administered.

Related Links

Interface on page 527
Protocol Version on page 218

Interface

Indicates whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN556B.

Valid Entry	Usage
network	User/network connection.
peer-master	Peer interface connection. Available only if QSIG Basic Call Setup is enabled

Side

Valid Entry	Usage
a b	Determines how glare conditions are handled for a peer-slave interface.

Related Links

Interface on page 214

ISDN-BRI Trunk Circuit Pack: page 2

Note:

If administering a TN2185 circuit pack, the system displays 8 ports; otherwise, the system displays 12 ports.

Note:

You cannot change the **Endpt Init**, **SPID**, or **Endpt ID** port parameters unless that port is busied out or unadministered. It is possible to change all other fields on this page even if the corresponding port is active.

If the **Interface** field on page 1 contains a valid value when the screen is submitted, the contents of the fields on page 2 for that port are validated. If the **Interface** field is blank when the screen is submitted, the fields on this page for that port reset to their default values.

Directory Number

The directory numbers assigned to the interface and allocated to two separate endpoints. This field must be administered in pairs. Accepts up to 10 characters.

Endpt ID

Valid Entry	Usage
00 to 62	The Endpoint Identifier expected by the far end. Communication Manager prevents changing this field unless the port is busied out or unadministered. Leading zeroes are significant and must not be ignored.

Endpt Init

Indicates whether the far end supports endpoint initialization. Communication Manager blocks you from changing this field unless the port is busied out or unadministered.

Valid Entry	Usage
y	Requires that an **SPID** be administered.
n	Requires that an **SPID** and **Endpt ID** not be administered.

Related Links

Endpt ID on page 528
SPID on page 529

Interworking Message

Determines what message Communication Manager sends when an incoming ISDN trunk call is routed over a non-ISDN trunk group.

Valid Entry	Usage
PROGress	Requests the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk. Normally-selected value.

Table continues…

Valid Entry	Usage
ALERTing	Causes the public network in many countries to play ringback tone to the caller. This value is used only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Max NCA TSC

Valid Entry	Usage
0 to 63	The maximum number of Non-Call-Associated Temporary Signaling Connections allowed on this BRI D-channel.

Port

The port number to which administered parameters apply.

SPID

The Service Profile Identifier (SPID) expected by the far end. Accepts up to 12 characters. Communication Manager prevents changing this field unless the port is busied out or unadministered. The only protocol supported for SPID initialization is Country Code 1. Trunks are not put in service if SPID installation is unsuccessful. Leading zeroes are significant and must not be ignored.

XID Test

If enabled, the far end supports the Layer 2 XID test.

Chapter 76: Language translations

The Language Translations screens display messages and their translations. Telephones display pre-translated messages from Communication Manager, which are available in the following languages:

- English
- French
- Italian
- Spanish

You can assign translations in any other language by using the Language Translations screens, but this method is deprecated. Instead, use Avaya Message Editing Tool (AMET) to assign translations for user-defined telephone messages.

If you use an editor other than AMET, the system does not update the Phone Message Files correctly and the files are not installed. For more details, see *Avaya Message Editing Tool (AMET) Job Aid*.

Note:

If you type user-defined in the Display Language field on the Station screen or the Attendant Console screen and messages are not defined for these screens, the system displays a string of asterisks on all display messages.

Warning:

If you use AMET, Communication Manager blocks access to the SAT user-defined translations-related screens. Therefore, do not use the Language Translations screens.

Warning:

If you do not use AMET, do not edit the user-defined translations-related screens through SAT.

You can download AMET from http://www.avaya.com.

Automatic Wakeup

Use this field to replace the English text for Automatic Wakeup messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Button labels

Use this field to replace the English text for Button labels that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Enhanced Abbreviated Dialing

Use this field to administer the Enhanced Abbreviated Dialing messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Leave Word Calling

Use this field to replace the English text for Leave Word Calling messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Malicious Call Trace

Use this field to replace the English text for Malicious Call Trace messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Miscellaneous call identifiers

Use this field to replace the English text for miscellaneous call identifiers that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536

Message 4 on page 535
Message 6 on page 536

Miscellaneous features

Use this field to replace the English text for miscellaneous features that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Property Management Interface

Use this field to replace the English text for property management interface messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Self Administration

Use this field to replace the English text for self administered messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Softkey Labels

Use this field to replace the English text for softkey labels that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

To provide unique labels for abbreviated dialing button types for softkey labels, Communication Manager replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D telephones.

On this screen, the digits following the **AD** are derived from the button position. If the first button is an AD button, then it is AD1 and the 15th button is AD15. All the AD buttons between 1 and 15 have the position number appended to AD.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Transfer Conference

Use this field to replace the English text for transfer conference messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536

Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

View Buttons

Use this field to replace the English text for buttons that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Vustats

Use this field to replace the English text for Vustats that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related Links

Message 12 on page 536
Message 15, 16, 18 on page 536
Message 4 on page 535
Message 6 on page 536

Message 4

The character "^" is a place holder.

English Text	Replacement Info
^-party conference in progress	"^" is replaced with the number of parties currently on the conference call.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Message 6

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to cancel or another line.	"^" is replaced with the letter of the line that is on soft hold.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Message 15, 16, 18

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Message 12

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Chapter 77: Listed Directory Numbers

With Listed Directory Numbers, Direct Inward Dialing (DID) numbers can be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

Example command: `change listed-directory-numbers`

Ext

Any valid extension number.

Name

The name used to identify the Listed Directory Number. Accepts up to 27 alphanumeric characters.

Night Destination

A valid assigned extension number that receives calls to these numbers when Night Service is active. Accepts up to thirteen digits.

TN

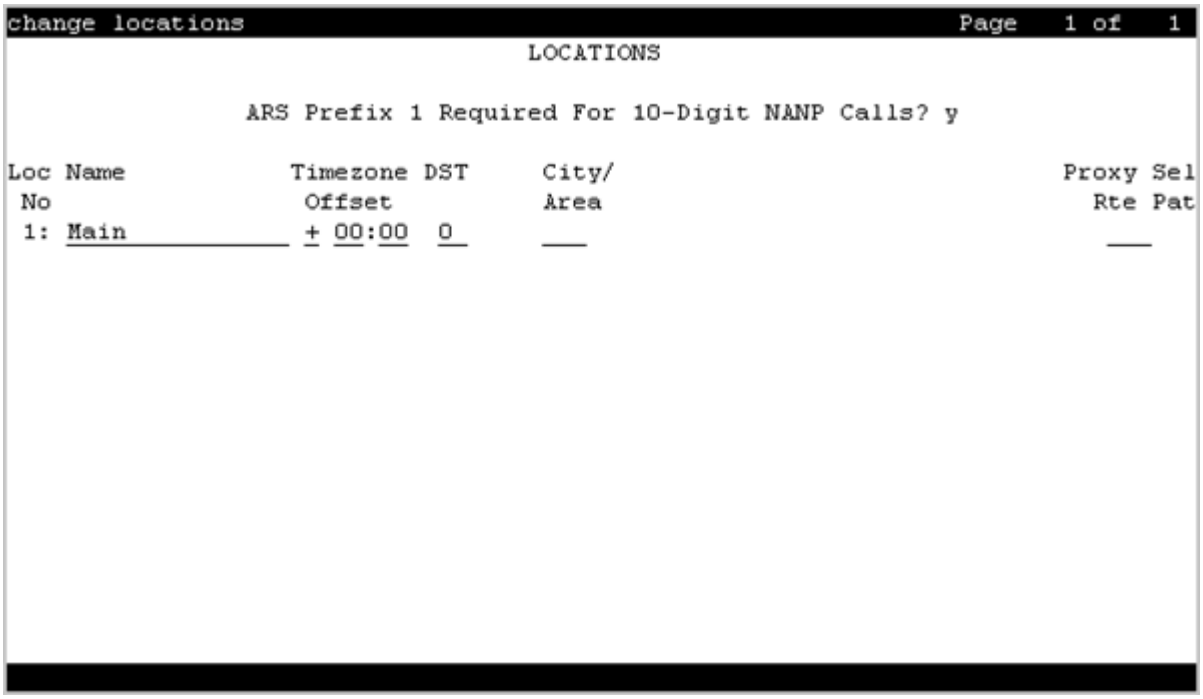
Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 78: Locations

This screen provides daylight saving time displays. You can set the area code for each location and administer different location information for each location. If the **Multiple Locations** feature is enabled, you can administer up to 2000 location specifications, depending on the configuration of Communication Manager. Otherwise, information for Location 1 applies to all locations.

You can use the following command to modify the fields on this screen:

```
change locations
```



ARS FAC

The Feature Access Code (FAC) for accessing Automatic Route Selection (ARS). Any valid FAC format is acceptable, up to four digits. Characters * or # are permitted, but only in the first position. Many locations are expected to share the same access code.

Related Links

Comments on this document? infodev@avaya.com

Feature Access Code (FAC) on page 239

ARS Prefix 1 Required for 10-Digit NANP Calls?

If enabled, a 1 must be dialed before all 10-digit NANP calls.

Attd FAC

The Feature Access Code (FAC) for connection to the attendant. Accepts up to two digits. Characters * or # are not permitted. Many locations are expected to share the same access code.

Available only if an Attendant Access Code has first been administered either for the Dial Plan or as a standard FAC.

Note:

Within a dial plan, FAC/DAC codes and extensions cannot both start with the same first digits. Either the FAC/DAC entries or the block of extensions must be changed to have a different first digit.

Related Links

Call Type on page 198
Attendant Access Code on page 241

Disp Parm

The administered display parameters for the location.

Related Links

Display Parameters on page 207

Loc Parm

Required when administering multiple locations.

Valid Entry	Usage
1 to 50 blank	The number of the corresponding Location Parameter set for this location. Default is blank.

Loc Number

Valid Entry	Usage
1 to 2000	The location number.

Name

A name you use for the location. Names are easier to remember than location numbers. Accepts up to 15 alphanumeric characters.

City/Area

The three-digit numbering plan area code for each location.

Prefix

Valid Entry	Usage
0 to 99999	Location prefix used with the Uniform Dialing Plan to identify the location of an originating call. All or part of the prefix is added to the front of the dialed string.

Related Links

Insert Digits on page 940

Proxy Sel Rte Pat

Valid Entry	Usage
1 to 999 blank	The routing pattern used to get to the proxy server.

Related Links

Route Pattern on page 651

DST

Required for each administered location.

Valid Entry	Usage
0	No Daylight Saving.
1 to 15 blank	The number for the Daylight-Saving Rule applied to this location.

Timezone Offset

Specifies how much time each location differs from the system time. Required for each administered location. Use +00:00 for the time zone offset for a single location media server.

Valid Entry	Usage
+	The time set on this location is a certain amount of time ahead of the system time.
-	The time set on this location is a certain amount of time behind the system time.

Valid Entry	Usage
0 to 23	The number of hours difference between this location and system time.

Valid Entry	Usage
0 to 59	The number of minutes difference between this location and system time.

Chapter 79: Location Parameters

Sets or changes certain administrable characteristics that determine part of a location's behavior. These include recall timing intervals and loss plans for two-party and conference calls.

Multiple instances of the Location Parameters screen are accessible if **Multiple Locations** is enabled for the system. If **Multinational Locations** is enabled, Location Parameters 2-50 contain the same fields as for Location Parameters 1. If **Multinational Locations** is disabled, the system does not display the following fields for Location Parameters 1:

- **Tone Generation Plan**
- **DCP Terminal-parameters Plan**
- **Country Code for CDR**

Location Parameters: page 1

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog telephones in the system.

Valid entry	Usage
1 to 25	The location country code. **Note:** This field must be set to 1 (US) for the Message Waiting Indicator on the Station to be set to neon.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entry	Usage
1 to 25	The location country code.

Companding Mode

The companding algorithm used by system hardware.

Valid Entry	Usage
A-Law	Generally used outside the U.S.
Mu-law	Generally used in the U.S.

Convert ISDN Public Numbers to International for Display

If you set the field to values other than ‘no’,Communication Manager converts the number to the international numbering format. The default value of this field is **no**.

Valid entries	Usage
All-stations	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on all configured stations.
All-digital	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on the configured digital stations.
DCP stations	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on the configured DCP and H.323 stations.
H.323 stations	Communication Manager adds a plus sign (+) to the calling or connected ISDN numbers that that appear on the H.323 stations.
No	Calling or connected ISDN numbers appear without the plus sign (+) on configured stations.

Country code for CDR

Available only if the Multinational Locations feature is enabled for the system.

Valid Entry	Usage
1 to 999	The country code to be used for Call Detail Recording information for a location. Default is 1.

Date Format on Terminals

The **Date Format on Terminals** field enables you to set the date format on the terminals, based on its location. The system displays this field only if you set the **Multinational location** field on the System Parameter Customer Options screen to y.

Valid Entry	Usage
mm/dd/yy	(month/day/year)
dd/mm/yy	(day/month/year)
yy/mm/dd	(Year/month/day)

DCP Terminal-parameters Plan

This field corresponds to the DCP terminal transmission parameters administered for the location. The system displays this field only if the Multinational Locations feature is enabled.

Valid entry	Usage
1 to 25	The terminal-parameters plan number. The default value is 1.

End OCM After Answer (msec)

If the **End OCM After Answer** field contains a non-blank value, Communication Manager starts the End OCM timer when Communication Manager receives an answer signal. The End OCM timer ensures that an outgoing call using OCM call classification is answered by an agent or an announcement within a specified time.

Valid Entry	Usage
100 to 25,000	The timeout value is in milliseconds. If the timer expires, Communication Manager disconnects the call classifier and connects the call to the administered intercept extension. If the call classifier classifies the call before the timer expires, the timer is cancelled and the call routed or treated appropriately.
blank	Indicates that the timer has no limit.

End of OCM Intercept Extension

Valid Entry	Usage
extension	The extension number Communication Manager connects the call to when the **End OCM After Answer** field expires. The extension number can be a recorded announcement, a vector directory number, or a hunt group extension.
blank	Indicates that the extension number is empty. If the **End OCM After Answer** field is set to a non-blank value, the **End of OCM Intercept Extension** field cannot be left blank.

International Access Code

An up to five-digit International Access Code. Default is blank.

Local E.164 Country Code

An up to three-digit E.164 Country Code. Default is blank.

For a list of country codes, see the *International Telecommunications Union List of ITU-T Recommendation E.164 Assigned Country Codes*.

Long Distance Access Code

Valid Entry	Usage
0 to 9	The long distance access code you want the system to prefix to the telephone number. Accepts up to five digits.
blank	Not administered. This is the default.

Off-PBX Feature Name Extension Set

Valid Entry	Usage
0 to 10 for a medium configuration 1 to 99 for a large configuration	Feature Name Extension (FNE) set that should be used for location based FNEs.
blank	Default FNE set is used. This is the default.

Tone Generation Plan

The value in this field corresponds to the tone generation characteristics administered for the location. The system displays this field only if the Multinational Locations feature is enabled for the system.

Valid Entry	Usage
1 to 25	The tone-generation plan number. The default value is 1.

RECALL TIMING

Disconnect Timing (msec)

Available only if a Flashhook Interval is not required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value is regarded as a disconnect.

Flashhook Interval

Enables or disables requiring a flashhook interval (recall window).

Forward Disconnect Timer (msec)

Valid Entry	Usage
25 to 1500 (in increments of 25)	Specifies the duration of a momentary disconnect sent by the server/switch to an analog station user when that user is the last party still off-hook on a call.

Lower Bound (msec)

Available only if a flashhook interval is required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval.

MF Interdigit Timer (sec)

Valid Entry	Usage
1 to 255	The maximum number of seconds Communication Manager waits for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires. This number must be less than the number of seconds administered for the short interdigit timer. Applies only to multifrequency signaling trunks.

Outgoing Shuttle Exchange Cycle Timer (sec)

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or none for the Multifrequency Signaling-Related System Parameters. This field applies only to multifrequency signaling calls made from Avaya Communication Manager.

Valid Entry	Usage
1 to 25	The number of seconds to time an exchange cycle (starts when the far end requests a digit until Avaya Communication Manager sends the requested digit.

Upper Bound (msec)

Available only if the flashhook interval is required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	The upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. A flash of 50 msec to 130 msec is

Valid Entry	Usage
	always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and is treated as the digit one.

Replace International Access Code with '+'

If you select values other than **no**, Communication Manager inserts a plus sign (+) instead of the international access code (IAC) when an international number appears on the designated endpoint. The default value of this field is **no**.

Valid entries	Usage
Usage	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on all configured stations.
All-digital	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on the configured digital stations.
DCP stations	Communication Manager adds the plus sign (+) to the calling or connected ISDN numbers that appear on the configured DCP stations.
H.323 stations	Communication Manager adds a plus sign (+) to the calling or connected ISDN numbers that that appear on the H.323 stations.
no	Calling or connected ISDN numbers appear without the plus sign (+) on configured stations.

Location Parameters: page 2

LOSS PLANS

2-Party Loss Plan/Tone Loss Plan

This field provides the default values for digital loss plan and for *n*-party conference loss.

Valid entry	Usage
1 to 25	A country code for the location.

Customize

Enables customization on the corresponding loss plan table. When **Customize** is enabled for the End-to-End total loss (dB) in a n-party conference, values can be changed by the administrator. When **Customize** is disabled, the End-to-End total loss (dB) in a n-party conference values are reset to the values that they would have had under the 2 Party Loss Plan. Available only if Digital Loss Plan Modification is enabled for the system.

End-to-End total loss (dB) in a n-party conference

Valid Entry	Usage
0 to 99	Provides total loss for a conference call with the designated number of parties. The higher the number listed for a call with a fixed number of parties, the more loss Communication Manager adds into a conference call with that number of parties; therefore, the conference call is quieter.

Note:

The End-to-End total loss for multi-party conference calls that is administered for this field is not always applied to a specific call.

Inter-location Loss Group

When inserting loss for a call, the server treats parties on the call who are in separate locations as if the location with the most parties were connected by an equal number of IP tie trunks as there are parties at other locations. The field specifies the digital loss group number that is used by these virtual IP tie trunks. The system displays this field only if the Multinational Locations feature is enabled.

Valid entry	Usage
1 to 19	The digital loss group number to use on inter-location calls involving this location. The default value is 18.

Location Parameters: page 3

FROM / TO

Valid Entry	Usage
–3 to 15	Identifies the variable digital loss values. An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Chapter 80: Login Administration

Beginning with Communication Manager 4.0, there is no longer a Login Administration screen. For details on screens used for login administration, see *Maintenance Commands for Avaya Aura® Communication Manager, branch Gateways and Servers*, 03-300431, and “AAA Services” in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Chapter 81: Logging levels

Administers logging of SAT activities. Specifies that commands associated with specific actions shown on this screen are logged by the system. The amount of detail to be logged is the same for all enabled actions and is specified by the **Log Data Values** field.

Note:

The defaults on this screen provide the same amount and type of logging as in Communication Manager releases prior to 4.0.

Example command: `change logging-levels`

Related Links

Log Data Values on page 550

Logging Levels: page 1

Enable Command Logging

If enabled, SAT activity is logged for selected commands.

Log Data Values

Valid Entry	Usage
none	Only the object, the qualifier, and the command action are logged.
new	The new value of any field is logged. The old value is not logged.
both	Both the prior field value and the field value after the change are logged.

Logging Levels: page 2

Log All Submission Failures

Form submission failures due to a security violation are always logged and are unaffected by this field.

Valid Entry	Usage
y	Record submission failures on the history log. An event is logged when Avaya Communication Manager rejects a form submission for any reason, such as an invalid entry in a field or a missing value.
n	Submission failures are not recorded on the history log.

Log CTA/PSA/TTI Transactions

Enables or disables logging transactions when extensions and physical telephones move between ports without additional administration from the administrator of Communication Manager.

If enabled, the system records Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Available only if Terminal Translation Initialization (TTI) is enabled for the system.

Related Links

Terminal Trans. Init. (TTI) on page 806

Log IP Registrations and Events

If enabled, the IP registrations can be logged in the history log.

Log PMS/AD Transactions

If enabled, the system records Property Management System (PMS) and Abbreviated Dialing (AD) events to the log.

Chapter 82: Loudspeaker Paging

The Loudspeaker Paging screen administers voice paging, deluxe voice paging, and chime paging.

Note:

To set up paging on a branch gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the Station screen. No entries on the Loudspeaker Paging screen are required.

Example command: `change paging loudspeaker`

CDR

If enabled, Communication Manager collects CDR data on the paging ports.

Code Calling — COR

Valid Entry	Usage
0 to 995	Assigns a Class of Restriction to a paging zone for chime paging.
blank	An unused paging zone.

Code Calling Playing Cycles

Valid Entry	Usage
1 to 3	The number of times a chime code plays when a user makes a chime page.
blank	Cannot be blank when administering chime paging (code calling).

Code Calling — TAC

Assigns a Trunk Access Code (TAC) to a paging zone for chime paging. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone used. Two zones cannot have the same TAC. A TAC in the zone designated ALL means that users can activate speakers in all the zones by dialing that code.

Valid Entry	Usage
1 to 4 digits	A Trunk Access Code (TAC) allowed by the dial plan.
*	Can be used as first digit.
#	Can be used as first digit.
blank	An unused paging zone.

Code Calling — TN

Valid Entry	Usage
1 to 100	Assigns a paging zone to a tenant partition for chime paging when Tenant Partitioning is in use.

Related Links

Tenant Partitioning on page 806

Location

Assigns a descriptive name for the physical location corresponding to each zone when Tenant Partitioning is in use. Accepts up to 27 characters.

Example

Typical entries might be “conference room A”, “warehouse”, or “storeroom”.

Port

Assigns a port on an auxiliary trunk circuit pack to a paging zone.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number

Table continues…

Valid Entry	Usage
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01–04 (Analog TIE trunks) 01–31	Six and seventh characters are the circuit number
1 to 250	Gateway
VI to V9	Module
01 to 31	Circuit
blank	An unused paging zone

Voice Paging — COR

Valid Entry	Usage
0 to 995	Assigns a Class of Restriction to a paging zone for voice paging.
blank	An unused paging zone.

Voice Paging — TAC

Use this field to assign a Trunk Access Code (TAC) to a paging zone for voice paging. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone. Two zones cannot have the same TAC. If you assign a TAC to the zone designated ALL, users can activate speakers in all the zones by dialing the code.

Valid entry	Usage
digits	Up to four digit TAC according to the dial plan
*	Can only be used as first digit
#	Can only be used as first digit
blank	An unused paging zone

Voice Paging Timeout (sec)

Limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages that are expected to broadcast and then add another 4 to 5 seconds.

Valid Entry	Usage
10 to 60 seconds	The maximum number of seconds any page lasts.
blank	The field cannot be blank when voice paging is administered.

Voice Paging — TN

Valid Entry	Usage
1 to 100	Assigns a paging zone to a tenant partition for voice paging if Tenant Partitioning is in use.

Related Links

Tenant Partitioning on page 806

Chapter 83: Maintenance-related system parameters

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431.

Chapter 84: Media-Gateway

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431.

Chapter 85: Message Waiting Indication Subscriber Number Prefixes

If the voice mail server is an AUDIX server connected using BX.25, you must fill in the Message Waiting Indication (MWI) Subscriber Number Prefixes to send or cancel a Message Waiting Indication to a served user. The served user is a user on the Private Telecommunication Network (PTN) to whom the MWI is sent on activation by the Message Center or Message Center PBX.

Using the QSIG MWI Supplementary Service, Message Center/Message Center PBX sends MWI Activate or MWI Deactivate message to the served user as a result of storage or retrieval of messages. Also, the served user is allowed to query the Message Center for MWI status.

Example command: `change isdn mwi-prefix`

AUDIX Mach ID

Valid entry	Usage
1 to 99	AUDIX number serving the switch of the AUDIX number field from the Switch Interface Administration screen.
blank	Available only if all the other records on the screen are blank.

Inserted Digits

Accepts up to 12 digits.

Valid entry	Usage
0 to 9	Digits prefixed to the AUDIX subscriber number to provide complete subscriber number to the Message Center PBX.
blank	No digits prefixed to the AUDIX subscriber number.

Machine ID

Available only if Machine ID is assigned to the qsig-mwi application in the Processor Channels Communication Interface.

Valid entry	Usage
1 to 20	Unique ID associated with each processor channel qsig-mwi application

Message Center ID

Accepts up to 20 digits. These are the Message Center ID digits to be sent a QSIG MWIActivate or MWIDeactivate message. It is necessary to enter the Message Center ID only if the system at the far-end of the QSIG connection needs it.

Routing Digits

Accepts up to four digits.

Valid entry	Usage
0 to 9, *, #	Digits comprising the AAR or ARS access code.
blank	No Routing Digits prefixed to the AUDIX subscriber number. For example, if UDP is used to route the call.

Send QSIG Message Center ID

Valid Entry	
y	Insert a message center ID in the MWI Activate and MWI Deactivate messages. For SIP-connected messaging servers, Communication Manager can be administered to insert a Message Center ID over QSIG connections to other servers by entering the Message Center ID, along with the Local Node Number for this Communication Manager server, in any record of this form.
n	Does not insert the message center ID in the MWI Activate and MWI Deactivate messages.

Chapter 86: Mode Code Related System Parameters

Establishes parameters associated with the Mode Code Voice Mail System Interface. Available only if the **Mode Code Interface** feature is enabled.

Example command: `change system-parameters mode-code`

Related Links

Mode Code Interface on page 292

MODE CODES (FROM SWITCH TO VMS)

Direct Dial Access-Trunk

The mode code that the media server or switch sends when an external caller dials the Voice Mail Sysem (VMS) access number. Accepts up to six digits. Also accepts #, *, and #00.

Direct Inside Access

The mode code that the media server or switch sends when a caller at an internal extension dials the Voice Mail System (VMS) access number. Accepts up to six digits. Also accepts #, *, and #00.

External Coverage

The mode code that the media server or switch sends when an external caller tries to reach a user at another extension and the call goes to the user's voice mail coverage. Accepts up to six digits. Also accepts #, *, and #00.

Internal Coverage

The mode code that Communication Manager sends when an internal caller tries to reach a user at another extension and the call goes to the user's voice mail coverage. Accepts up to six digits. Also accepts #, *, and #00.

Refresh MW Lamp

The mode code that Communication Manager sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps. Accepts up to six digits. Also accepts #, *, and #00.

System In Day Service

This value is used by the Voice Mail System (VMS) to indicate that Communication Manager has changed from Night to Day Service. Accepts up to six digits. Also accepts #, *, and #11.

System In Night Service

This value is used by the Voice Mail System (VMS) to indicate that Communication Manager has changed from Day to Night Service. Accepts up to six digits. Also accepts #, *, and #12.

OTHER RELATED PARAMETERS

DTMF Duration On

Valid Entry	Usage
75 to 500 — in multiples of 25	The duration in milliseconds of mode code digits sent to the VMS. This field cannot be blank.

Off

Valid Entry	Usage
Between 75 and 200 — in multiples of 25	The pause in milliseconds between mode code digits as they are sent to the VMS. This field cannot be blank.

Remote VMS Extensions- First

The first remote UDP VMS hunt group extension. Available only if **Mode Code for Centralized Voice Mail** is enabled for the system.

Related Links

Mode Code for Centralized Voice Mail on page 802

Remote VMS Extensions - Second

The second remote UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension. Available only if **Mode Code for Centralized Voice Mail** is enabled for the system.

Related Links

Mode Code for Centralized Voice Mail on page 802

Sending Delay

Valid Entry	Usage
75 to 1000 — in multiples of 25	The delay in milliseconds between the time answer supervision is received from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

The extension of a hunt group containing VMI extensions. A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Chapter 87: Modem Pool Group

There are two types of conversion resources for Modem Pooling. The first type, an integrated conversion resource, is a circuit pack that emulates a Trunk Data Module connected to a 212A-type modem. Two conversion resources are on each circuit pack.

The second type, a combined conversion resource, is a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource can be either a Modular Trunk Data Module (MTDM) or 7400A Data Module and connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port.

Note:

The **Speed**, **Duplex**, and **Synchronization** fields cannot be filled out for the “integrated” pooled modem screens but can be assigned on the “combined” pooled modem screen. The integrated conversion resource automatically will adjust its speed and synchronization to the endpoint it is connected to. In synchronous mode, the integrated modem pool can operate at 1200 baud. In asynchronous mode, it can operate at 300 or 1200 baud. Full-duplex operation is always used.

Example command: `change modem-pool` *n*, where *n* is the modem pool number.

Answer Supervision Timeout (sec)

Available only with a combined type modem pool.

Valid Entry	Usage
1 to 255	The number of seconds to wait before the far-end answers.
0	No answer supervision

Related Links

Group Type on page 564

CF-CB Common

If enabled, the CF and CB leads on the conversion resource are logically connected. Available only with an integrated type modem pool.

Related Links

Group Type on page 564

Direction

The direction of the call for which the modem pool operates. Available only with a combined type modem pool.

Valid Entry	Usage
incoming	Converts an analog signal to digital for the data endpoint
outgoing	Converts analog to digital (or digital to analog) for data calls
two-way	Allows incoming and outgoing data communication

Related Links

Group Type on page 564

Duplex

The duplex mode of the conversion resources in the group. Entry required for a combined type modem pool.

Valid Entry	Usage
full	Can talk and listen at the same time
half	Cannot talk and listen at the same time

Related Links

Group Type on page 564

Group Number

The modem pool group number.

Group Type

The type of physical model pool.

Valid Entry	Usage
integrated	Maps to the Pooled Modem circuit pack
combined	Maps to an external modem pool when a data module and a modem exists

Hold Time (min)

Valid Entry	Usage
1 to 99	The maximum number of minutes that a conversion resource in the group can be held while a call waits in a queue or reserved after Data Call Preindication.

Loss of Carrier Disconnect

If enabled, permits conversion resource to disconnect if it detects a dropped carrier.

Available only with an integrated type modem pool.

Related Links

Group Type on page 564

Modem Name

The name of the modem pool. Accepts from one to six characters. Available only with a combined type modem pool.

Related Links

Group Type on page 564

Receive Space Disconnect

If enabled, the conversion resource can disconnect after receiving 1.6 seconds of space.

Available only with an integrated type modem pool.

Related Links

Group Type on page 564

Receiver Responds to Remote Loop

If enabled, the far-end modem can put conversion resource into loop back mode.

Available only with an integrated type modem pool.

Related Links

Group Type on page 564

Send Space Disconnect

If enabled, the conversion resource can send 4 seconds of space before disconnecting

Available only with an integrated type modem pool.

Related Links

Group Type on page 564

Speed

Valid Entry	Usage
LOW (0 to 300 blind sampled) 300 1200 2400 4800 9600 19200	The communication speed in bits per second of the conversion resources in the group. One to three speeds are separated by slashes. For example, 300/1200/2400 indicates a maximum of three running speeds. Entry is required for a combined type modem pool.

Related Links

Group Type on page 564

Synchronization

The synchronization mode of the conversion resources in the group. Entry required for a combined type modem pool.

Valid Entry	Usage
sync	Synchronous
async	Asynchronous

Related Links

Group Type on page 564

Time Delay

Available only with a combined type modem pool.

Valid Entry	Usage
0 to 255	The time delay in seconds to insert between sending the ringing to the modem and the off-hook alert to the data module.

Related Links

Group Type on page 564

CIRCUIT PACK ASSIGNMENTS

Circuit Pack Location

The port associated with the conversion resource on the integrated modem pool circuit pack. Available only with an integrated type modem pool.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Example

01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Related Links

Group Type on page 564

PORT PAIR ASSIGNMENTS

Analog Digital

The port numbers of the modem/TDM pair in a conversion resource. Two port entries are required. Available only with a combined type modem pool.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Example

01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

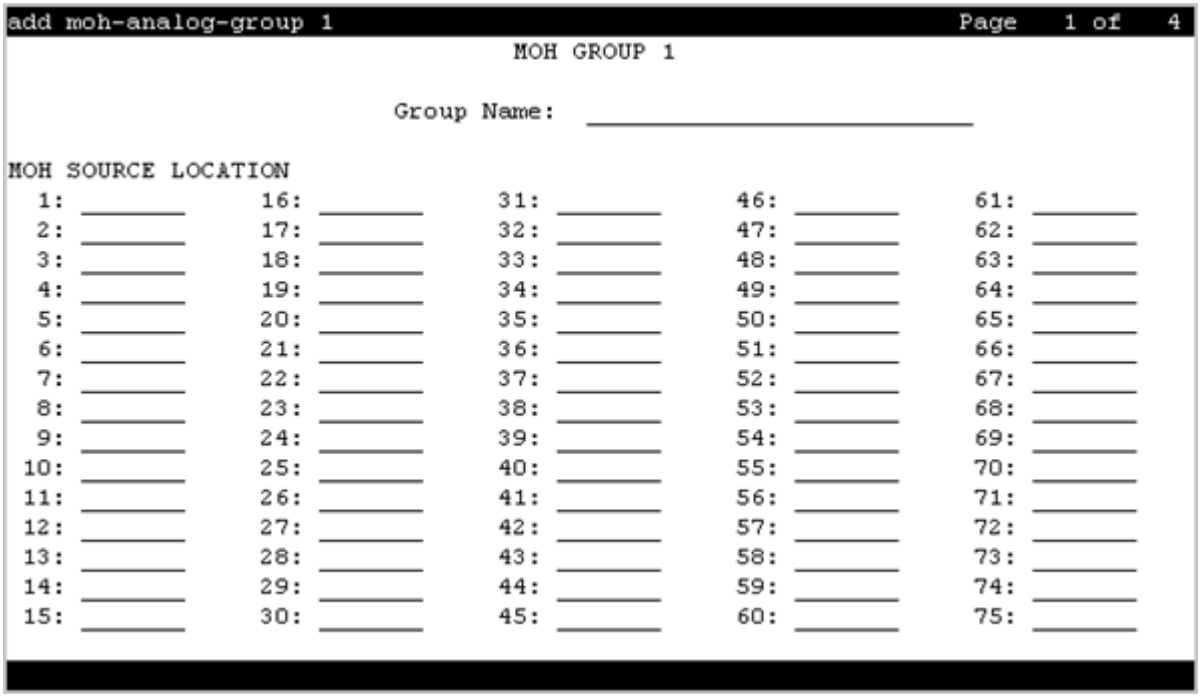
Related Links

Group Type on page 564

Chapter 88: MOH Group

Use the MOH Group screen to define a collection of analog station and/or aux trunk port circuit pack ports that are connected to external audio sources for use with the Music on Hold feature.

Example command: `change moh-analog-group`



MOH Source Location

The Music-on-hold analog or aux-trunk port location

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number

Table continues…

Valid Entry	Usage
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Group Name

The name that identifies the Music-on-hold (MOH) group.

Chapter 89: Multifrequency-Signaling-Related Parameters

Sets the system or location parameters associated with multifrequency signaling.

Note:

With the Multinational Locations feature enabled, you can assign MFC signal sets per trunk group, rather than system-wide.

Example command: `change multifrequency-signaling`

Related Links

Multinational Locations on page 803

Multifrequency-Signaling-Related Parameters: page 1

ANI Prefix

The prefix to apply to an extension when ANI is sent to the local telephone company central office. Accepts one to six digits.

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Related Links

Outgoing Call Type on page 575

Backward Cycle Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds the system waits to send the check frequency after receiving an MFE signal.

Related Links

Incoming Call Type on page 573

Collect All Digits Before Seizure

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
y	The system collects all the digits before seizing the trunk. Values administered for **ANI Req** for AAR and ARS Digit Conversion do not apply.
n	ANI collection is controlled by ARS administration.

Related Links

ANI Reqd

Outgoing Call Type on page 575

Convert First Digit End-of-Dial To

The digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Available only if **Private Group II Permissions and Public Interworking** is enabled.

Related Links

Private Group II Permissions and Public Interworking on page 577

Default ANI

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
2 to 15 digits	The switch or server identification number that is sent to the local telephone company central office (CO) when ANI is requested (by the CO) on a particular call but is unavailable, such as on tandem tie trunk calls.
blank	Use for tandem switching. If blank, administer a value to send for ANI on outgoing calls when it is unavailable.

Related Links

ANI Not Available on page 582

Outgoing Call Type on page 575

Comments on this document? infodev@avaya.com

Forward Cycle Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds the system waits to receive the check frequency after sending an MFE signal. Communication Manager drops the call if the time runs out before it receives check frequency.

Related Links

Incoming Call Type on page 573

Group II Called Party Category

The type of Group II signals that should be used on the outgoing R2-MFC call. Available only if the **Outgoing Call Type** is group-ii-mfc and **Use COR for All Group II Responses** is disabled.

Valid Entry	Usage
user-type	The type of telephone making the call determines the type of Group II signal that the switch or server sends (normal = ordinary telephone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of Group II signal that the server sends.

Related Links

Outgoing Call Type on page 575
Use COR for All Group II Responses on page 578

Incoming Call Type

Valid Entry	Usage
group-ii-mfc non-group-ii-mfc	The signal type that a local telephone company central office uses to place an incoming call to the server.
mfe	Multi-frequency Espanol used for Spain

Incomplete Dial Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
45 to 255	The number of seconds Communication Manager waits from the start of a call until the end of the check frequency of the last signal. Communication Manager drops the call if the time runs out before it receives the check frequency.

Related Links

Incoming Call Type on page 573

Maintenance Call Type

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.

Valid Entry	Usage
1	The Belgium maintenance sequence is indicated when the local telephone company central office (CO) sends an MFC maintenance tone.
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.
none	Not administered.

Related Links

Incoming Call Type on page 573

Maximum Resend Requests

The threshold number of resend type MFC signals the server running Communication Manager accepts during an outgoing call.

Valid Entry	Usage
1 to 99	The number of resend requests.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

MF Signaling Intercept Treatment - Incoming

Valid Entry	Usage
y	Sends the group B signal for the intercept to the local telephone company central office and play intercept tone on the trunk.
n	Uses normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Defines the treatment for outgoing calls that cannot be completed as dialed. Available only if **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Requires entry of the extension number for the announcement.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

Related Links

Outgoing Call Type on page 575

MFE Type

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
2/5 2/6	Determines which public signaling Communication Manager uses.

Related Links

Incoming Call Type on page 573

Outgoing Call Type

Valid Entry	Usage
group-ii-mfc	The signal type that the switch or server uses to place an outgoing call to a local telephone company central office.
mfe	Multi-frequency Espanol used in Spain.
none	Not administered.

Outgoing Forward Signal Absent Timer (sec)

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
11 to 255	The maximum time to elapse between forward signals on outgoing calls. The timer starts and restarts when a forward tone is taken off the link, and it stops when the next forward tone is applied to the link.

Related Links

Outgoing Call Type on page 575

Outgoing Forward Signal Present Timer (sec)

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
1 to 255	The maximum time to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when Communication Manager begins sending a forward signal and stops when Communication Manager receives the backward signal.

Related Links

Outgoing Call Type on page 575

Outgoing Start Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Related Links

Incoming Call Type on page 573

Overlap Sending on Link-to-Link Tandem Calls

If enabled, and calls are tandemed between servers, then ANI for the switch or server is sent to the terminating switch if that switch requests ANI before Communication Manager receives it from the originating server or switch. The terminating server or switch can request ANI before the receipt of the last address digit if it is not running Communication Manager, or if Communication Manager is administered to request the call category at the start of the call.

If enabled, Communication Manager sends and receives digits one digit at a time instead of enbloc. With enbloc, digits are not sent until the entire group of digits is received.

Available only if **Collect All Digits Before Seizure** is disabled.

Related Links

Collect All Digits Before Seizure on page 572
Request Call Category at Start of Call on page 581

Private Group II Permissions and Public Interworking

If enabled, then Communication Manager:

- Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls.
- Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls.
- Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls.
- Checks station permissions for call forward off-net calls.

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or none.

Related Links

Incoming Call Type on page 573
Outgoing Call Type on page 575

Received Signal Gain (dB)

Valid Entry	Usage
-15 to 3	The number for the loss or gain when the MFC port listens to the trunk port. Communication Manager listens with a range of -5 to -35. This value moves the range (for example, a value of -5 provides a range of -10 to -40). Also applies to Russian MF Shuttle trunks.

Request Incoming ANI (non-AAR/ARS)

If enabled, ANI is requested on incoming R2-MFC calls.

Available only if **Incoming Call Type** is group-ii-mfc or mfe and the **Outgoing Call Type** is group-ii-mfc or mfe. Applies only if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Related Links

Incoming Call Type on page 573
Outgoing Call Type on page 575

Transmitted Signal Gain (dB)

Valid Entry	Usage
-15 to 3	The number for the loss or gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE. This field adds gain or loss to the starting value of -5. Also applies to Russian Shuttle trunks and Russian multi-frequency ANI.

Use COR for All Group II Responses

If enabled, the COR administered category is used for both the calling party and called party categories.

Available only if the **Outgoing Call Type** is group-ii-mfc.

Related Links

Outgoing Call Type on page 575

Use COR for Calling Party Category

Indicates the category to send with ANI if requested on an outgoing R2-MFC call. Available only if the **Outgoing Call Type** is group-ii-mfc and **Use COR for All Group II Responses** is disabled.

Valid Entry	Usage
y	The calling facility's COR is used to determine category.
n	The calling party's user-type COR is used to determine category.

Related Links

Outgoing Call Type on page 575
Use COR for All Group II Responses on page 578

NEXT ANI DIGIT

Related Links

Incoming Call Type on page 573
Outgoing Call Type on page 575

Incoming

Available only if the **Incoming Call Type** is group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
next-digit next_ani_digit send-ani	Determines whether the Next ANI Digit signal is the same as the send-ani signal, the next-digit signal, or another signal defined as next_ani_digit.

Outgoing

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
next-digit next_ani_digit send-ani	Determines whether the Next ANI Digit signal is the same as the send-ani signal or the next-digit signal or another signal defined as next_ani_digit.

Related Links

Outgoing Call Type on page 575

Multifrequency-Signaling-Related Parameters: page 2

These fields define call category and ANI information. For India, the ANI can be requested without the call category information.

Address Digits Include End-of-Digits Signal

If enabled, indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the local telephone company central office for outgoing calls.

ANI Source for Forwarded & Covered Calls

Valid Entry	Usage
caller	Send the calling party's ANI when calls are redirected.
forwarder	Send the forwarding party's ANI when calls are redirected.

Call Category for Vector ii-digits

If enabled, allows the use of the call category digit, which is part of ANI, as the ii-digits on call vector steps.

Do Not Send Group B Signals to CO

Available only if the **Incoming Call Type** is group-ii-mfc.

Valid Entry	Usage
y	Does *not* send Group-B signals to complete an incoming call.
n	Sends Group-B signals to complete an incoming call.

Related Links

Incoming Call Type on page 573

Number of Incoming ANI Digits

Valid Entry	Usage
0 to 15	The number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

Valid Entry	Usage
0 to 15	The number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls - Group I, Avaya Communication Manager appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected. Applies to Russian shuttle trunks, and MFC and MFE trunks.

Outgoing II by COR

Available only if either **Use COR for Calling Party Category** or **Use COR for All Group II Responses** are enabled.

Valid Entry	Usage
1 to 10	The Group II signal sent to the local telephone company central office on outgoing calls can be administered per COR (Class of Restriction) and per trunk group. The Group II signal is administered per COR. That per-COR value in turn can be mapped into a different outgoing signaling parameter set. The values for this field administer that outgoing mapping.

Related Links

Use COR for All Group II Responses on page 578
Use COR for Calling Party Category on page 578

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information is sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

If enabled, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Communication Manager sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Communication Manager sends the Send-ANI backward signal upon receipt of the first address signal.

Request CPN at Start of Call

With Request CPN at Start of Call, Communication Manager can collect ANI and call category immediately after receipt of the first address digit.

If enabled, provides ANI Calling Party Number (CPN) and call category immediately after receiving the first address digit.

Available only if the **Incoming Call Type** is group-ii-mfc.

Related Links

Incoming Call Type on page 573

Restart ANI from Caller Category

If enabled, Avaya Communication Manager sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Truncate Station Number in ANI

Applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid Entry	Usage
beginning ending no	Defines the side of the extension number from which to truncate when station ANI is sent to the local telephone company central office and the combined length of the ANI prefix and extension number is greater than the administered Number of Outgoing ANI

Valid Entry	Usage
	Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if ANI for switch or server is sent.

Related Links

Number of Outgoing ANI Digits on page 580

INCOMING / OUTGOING

ANI Available

Valid Entry	Usage
1 to 15 blank	The signal number used for incoming ANI-Available.

ANI Not Available

Required if the **Default ANI** is not administered.

Valid Entry	Usage
1 to 15 blank	The signal number used for outgoing ANI-Available. Communication Manager outpulses the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Communication Manager outpulses the ANI-Not-Available forward signal to the local telephone company central office on outgoing calls where ANI is not possible.

Multifrequency-Signaling-Related Parameters: page 3

These fields define the meaning of MFC tones for calls originated at the local telephone company central office.

When the system initially displays the screen, either of two sets of default values is possible. One set is for the Group II call type; the other set is for non-Group II call type. In each set, the default value for each field is set to the most common value.

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.

Related Links

Incoming Call Type on page 573

INCOMING FORWARD SIGNAL TYPES (Tones from CO)

Group I

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group I signal type can be administered for each code.

Group I signals are a set of forward signals generated by the originating server.

Valid Entry	Usage
ani-avail	Used in Hungary. If this signal is defined and Automatic Number Identification (ANI) is requested on outgoing R2-MFC calls, ANI is sent to the local telephone central office before ANI caller digits are sent. This signal is sent after the ANI caller category signal. Available if **Incoming Call Type** is group-ii-mfc.
ani-not-avail	Used on Direct Outward Dialing calls in Brazil and Columbia. Communication Manager sends this signal to the local telephone company central office when it receives an Automatic Number Identification (ANI) request and the caller's number is unavailable. Available if **Incoming Call Type** is group-ii-mfc.
drop	When this signal is received from the local telephone company central office, Avaya Communication Manager starts the disconnect sequence and drops the call. Available if **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.
end-of-ani	This signal is used on Direct Outward Dialing and Direct Inward Dialing calls. Communication Manager sends this signal to indicate the end-of-ANI digits when Automatic Number Identification (ANI) digits are sent to the local telephone company central office. Available if **Incoming Call Type** is group-ii-mfc.
end-of-dial	This signal is used when open numbering is used on Direct Inward Dialing calls. The local telephone company central office sends this signal to indicate the end-of-dial digits and Communication Manager responds with a request for a Group II signal. Available if **Incoming Call Type** is group-ii-mfc.
ignored	If this signal is received from the local telephone company central office, Communication Manager sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged. Available if **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.
maint-call	The local telephone company central office (CO) sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO. This signal can be used on Direct Inward Dialing calls in Saudi Arabia. Available if **Incoming Call Type** is group-ii-mfc.

Table continues…

Valid Entry	Usage
send-congest	When Communication Manager receives this signal from the local telephone company central office (CO) on a Direct Inward Dialing call, it returns a congestion signal (Group A), in compel (not pulse) mode, to the CO. Available if **Incoming Call Type** is group-ii-mfc.

Group II

Displays message codes 1 to 15. A Group II signal type can be administered for each code.

Group II signals are a more elaborate set of forward signals generated by the originating server.

Valid Entry	Usage
attendant	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant regardless of the extension dialed. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.
busy-rt-attd	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.
data-call	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
data-verify	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls and the terminating extension is not a data extension, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
drop	When this signal is received from the local telephone company central office (CO), Communication Manager starts the disconnect sequence and drops the call.
maint-call	The local telephone company central office (CO) sends a signal to indicate that a call is a maintenance call and Avaya Communication Manager prepares the special maintenance call sequences for the CO.
normal	This signal indicates that the caller is a normal subscriber. If it is received on a Direct Inward Dialing (DID) call, the call is terminated at the called extension. For an outgoing MF signaling call that uses Group II signaling, this signal is sent to the local telephone company central office (CO) when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and Direct Outward Dialing calls.
send-intercept	If Communication Manager receives this signal from the local telephone company central office (CO) on a Direct Inward Dialing call, it returns Group B intercept signal to the CO.

Table continues…

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Valid Entry	Usage
toll-auto	This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the local telephone company central office. A special busy signal can be defined by choosing the option toll-busy on the incoming Group B signals.
toll-operator	This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

INCOMING BACKWARD SIGNAL TYPES (Tones to CO)

Group A

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group A signal type can be administered for each code.

Group A signals are backward signals generated by the destination server or switch.

Valid Entry	Usage
congestion	The local telephone company central office sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.
end-of-dial	This signal is sent to indicate the end of the address digit string. For MF Group II calls, this signal requests a Group II signal and switches the sender over to the Group B signaling mode. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
intercept	The local telephone company central office sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.
next-ani-digit	Communication Manager sends this signal to request the next ANI digit. This signal is used on Direct Inward Dialing and Direct Outward Dialing calls.
next-digit	Communication Manager sends this signal to request the next digit. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
send-ani	The local telephone company central office (CO) sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on Direct Inward Dialing (DID) calls. This signal is used on both Direct Outward Dialing and DID calls.
setup-sppath	The local telephone company central office sends this signal to Communication Manager to set up a speech path. This signal is used on Direct Outward Dialing calls and on Direct Inward Dialing calls in Belgium.

Group B

Displays message codes between 1 and 15. A Group B signal type can be administered for each code

Group B signals enhance Group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses Group II signals, the destination end answers with Group B signals.

Not available if **Do Not Send Group B Signals to CO** is enabled.

Valid Entry	Usage
busy	This signal is sent to indicate that the called party is busy. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office if there is no coverage point to terminate the call. If Communication Manager receives this signal on Direct Outward Dialing calls, it plays busy tone to the calling party and drops the trunk.
congestion	This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that a resource is not available. On Direct Outward Dialing calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.
free	This signal indicates that the called party is idle. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on Direct Outward Dialing calls, it connects the trunk to the calling party.
intercept	This signal indicates that the called party number is not in service or is incorrect. On Direct Inward Dialing calls, if intercept treatment is set to provide a tone, tone is sent to the local telephone company central office to indicate that the called number is invalid. If Communication Manager receives the signal on Direct Outward Dialing calls, it plays intercept tone to the calling party and drops the trunk.
mct	This signal identifies the call as one that needs to be traced by the local telephone company central office (CO). Avaya Communication Manager then generates an MFC Call Trace Backward Signal administered for multifrequency signaling during call setup instead of the “free” signal. If the terminating station’s Class of Restriction (COR) is enabled, the CO collects trace information before releasing the calling party. If the station’s COR has MF Incoming Call Trace enabled and the “mct” signal is not defined, then the “free” signal is sent.
tariff-free	This signal is sent when the trunk group provides an 800 service. Avaya Communication Manager generates an MFC tariff-free backward signal during call setup instead of the "free" signal, facilitating local telephone company central office billing. If the trunk is administered as a tariff-free trunk and the “tariff-free” signal is not defined, then the “free” signal is sent.
tie-free	This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.
toll-busy	This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Related Links

Do Not Send Group B Signals to CO on page 580

Multifrequency - Signaling- Related Parameters: page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the switch or server.

The system displays this screen only if **Outgoing Call Type** is group-ii-mfc or mfe.

Related Links

Outgoing Call Type on page 575

OUTGOING FORWARD SIGNAL TYPES (Tones to CO)

Group I

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group I signal type can be administered for each code.

Group I signals are a set of forward signals generated by the originating server.

Valid Entry	Usage
ani-avail	Used in Hungary. If this signal is defined and Automatic Number Identification (ANI) is requested on outgoing R2-MFC calls, ANI is sent to the local telephone central office before ANI caller digits are sent. This signal is sent after the ANI caller category signal.
ani-not-avail	Used on Direct Outward Dialing calls in Brazil and Columbia. Communication Manager sends this signal to the local telephone company central office when it receives an Automatic Number Identification (ANI) request and the caller's number is unavailable.
end-of-ani	This signal is used on Direct Outward Dialing and Direct Inward Dialing calls. Communication Manager sends this signal to indicate the end-of-ANI digits when Automatic Number Identification (ANI) digits are sent to the local telephone company central office.
end-of-digits	This signal is sent by the originating server that makes outgoing calls, sends digits, and receives a next-digit Group A signal from the destination server or switch when there are no more digits to be sent. This signal is also sent when Communication Manager does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani Group A signal from the destination end when there are no more Automatic Number Identification (ANI) digits to be sent. If both end-of-digits and end-of-ani are assigned, Communication Manager uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

Group II

Displays message codes between 1 and 15. A Group II signal type can be administered for each code. Each entry can only appear once in the Group II column.

Group II signals are a more elaborate set of forward signals generated by the originating server.

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Valid Entry	Usage
attendant	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant regardless of the extension dialed. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.
data-call	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
normal	This signal indicates that the caller is a normal subscriber. If it is received on a Direct Inward Dialing (DID) call, the call is terminated at the called extension. For an outgoing MF signaling call that uses Group II signaling, this signal is sent to the local telephone company central office (CO) when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and Direct Outward Dialing calls.
toll-auto	This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the local telephone company central office. A special busy signal can be defined by choosing the option toll-busy on the incoming Group B signals.

OUTGOING BACKWARD SIGNAL TYPES (Tones from CO)

Group A

Displays message codes between 1 and 15. A Group A signal type can be administered for each code.

Group A signals are backward signals generated by the destination server or switch.

Valid Entry	Usage
congestion	The local telephone company central office sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.
drop	When this signal is sent, the receiving end starts the disconnect sequence.
end-of-dial	This signal is sent to indicate the end of the address digit string. For MF Group II calls, this signal requests a Group II signal and switches the sender over to the Group B signaling mode. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
intercept	The local telephone company central office sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.

Table continues…

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Valid Entry	Usage
last-2-digits	Communication Manager sends this signal to adjust the outpulsing pointer so that the last three digits can be resent. This signal is used on Direct Outward Dialing calls.
last-3-digits	Communication Manager sends this signal to adjust the outpulsing pointer so that the last four digits can be resent. This signal is used on Direct Outward Dialing calls.
last-digit	Communication Manager sends this signal to adjust the outpulsing pointer so that the last two digits can be resent. This signal is used on Direct Outward Dialing calls.
next-ani-digit	Communication Manager sends this signal to request the next ANI digit. This signal is used on Direct Inward Dialing and Direct Outward Dialing calls.
next-digit	Communication Manager sends this signal to request the next digit. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
restart	Communication Manager sends this signal to request the whole digit string again. This signal is used on Direct Outward Dialing calls.
resend-digit	Communication Manager sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on Direct Outward Dialing calls.
send-ani	The local telephone company central office (CO) sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on Direct Inward Dialing (DID) calls. This signal is used on both Direct Outward Dialing and DID calls.
setup-sppath	The local telephone company central office sends this signal to Communication Manager to set up a speech path. This signal is used on Direct Outward Dialing calls and on Direct Inward Dialing calls in Belgium.

Group B

Displays message codes between 1 and 15. A Group B signal type can be administered for each code.

Group B signals enhance Group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses Group II signals, the destination end answers with Group B signals.

Valid Entry	Usage
busy	This signal is sent to indicate that the called party is busy. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office if there is no coverage point to terminate the call. If Communication Manager receives this signal on Direct Outward Dialing calls, it plays busy tone to the calling party and drops the trunk.
congestion	This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that a resource is not available. On Direct Outward Dialing calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.
free	This signal indicates that the called party is idle. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that the called

Table continues…

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Valid Entry	Usage
	party is idle and the call is terminated successfully. If Communication Manager receives this signal on Direct Outward Dialing calls, it connects the trunk to the calling party.
intercept	This signal indicates that the called party number is not in service or is incorrect. On Direct Inward Dialing calls, if intercept treatment is set to provide a tone, tone is sent to the local telephone company central office to indicate that the called number is invalid. If Communication Manager receives the signal on Direct Outward Dialing calls, it plays intercept tone to the calling party and drops the trunk.
mct	This signal identifies the call as one that needs to be traced by the local telephone company central office (CO). Avaya Communication Manager then generates an MFC Call Trace Backward Signal administered for multifrequency signaling during call setup instead of the “free” signal. If the terminating station’s Class of Restriction (COR) is enabled, the CO collects trace information before releasing the calling party. If the station’s COR has MF Incoming Call Trace enabled and the “mct” signal is not defined, then the “free” signal is sent.
tariff-free	This signal is sent when the trunk group provides an 800 service. Avaya Communication Manager generates an MFC tariff-free backward signal during call setup instead of the "free" signal, facilitating local telephone company central office billing. If the trunk is administered as a tariff-free trunk and the “tariff-free” signal is not defined, then the “free” signal is sent.
toll-busy	This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Chapter 90: Multiple Level Precedence & Preemption (MLPP) Parameters

Use this screen to set up system parameters for the Multiple Level Precedence & Preemption feature.

Example command: `change system-parameters mlpp`

ANNOUNCEMENTS

ASAC Enabled

Note:

The **ASAC Enabled** field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Assured Services is a U.S. military term for making high-precedence calls succeed, even if that involves blocking or preempting low-precedence calls. Assured Services Admission Control (ASAC) prevents precedence calls from connecting to the caller if the caller is already on a higher or an equal precedence call. For example, if the system is at the budget limit and all calls that are connected are Flash Override calls, and another Flash Override call tries to connect, ASAC blocks the call.

This field is available only when you set the **Multiple Level Precedence and Preemption** field on the System Parameters Customer Options screen to `y`.

Valid entry	Usage
y	Makes all ASAC-related fields available on the SIP Trunk Group screen. ASAC also allows Destination Code Control (DCC) functionality.
n	Makes ASAC-related fields unavailable on the SIP Trunk Group screen. Therefore, the ASAC functionality is unavailable for calls. The default value is `n`.

Attendant Queue Announcement

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The extension of the attendant queue announcement for incoming precedence calls that are above the Routine precedence level and are placed in the attendant waiting queue.

To enable **Attendant Queue Announcement**, you must set the **Annc Type** field on the Announcements/Audio Sources screen to `Integrated`.

Blocked Precedence Level

The extension of the Blocked Precedence Level announcement.

Busy, Not Equipped

The extension of the Busy, Not Equipped for Preemption announcement.

Service Interruption

The extension of the Service Interruption announcement.

Unauthorized Precedence Level

The extension of the Unauthorized Precedence Level announcement.

Vacant Code

The extension of the Vacant Code announcement.

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

Caution:

Do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits are different across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

Attendant Diversion Timing (sec)

Valid Entry	Usage
10 to 99 blank	Controls how many seconds this type of call rings before the call is routed to the Remote Attendant Route String that consists of any valid telephone number on the network. Attendant Diversion Timing is usually a backup answering position for the remote attendant console. The Remote Attendant Route String does not raise the precedence level of the call.

Default Route Digit

Available only if Worldwide Numbering Dial Plan is enabled. A valid digit is required in this field.

Valid Entry	Usage
0	Voice call (the default value)
1	Circuit switched data call
2	Satellite avoidance call
3	(reserved)
4	(reserved)
5	Hotline voice grade call
6	Hotline data grade call
7	(reserved)
8	(reserved)
9	(reserved)

Related Links

Worldwide Numbering Dial Plan Active on page 596

Default Service Domain

Valid Entry	Usage
0 to 16777215	The system service domain number. This number must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

Flash

The digit assignment for Flash precedence level calls. The default is 1.

Flash Override

The digit assignment for Flash Override precedence level calls. The default is 0.

Immediate

The digit assignment for Immediate precedence level calls. The default is 2.

ISDN Precedence Call Timeout (sec)

Valid Entry	Usage
4 to 30	The timeout seconds used instead of the Precedence Call Timeout when the call is from a Multiple Level Precedence and Preemption (MLPP) ISDN-PRI trunk. Default is 30 seconds.

Line Load Control Restriction Level

Determines what stations, based on their COR, are restricted from originating calls.

Valid Entry	Usage
0	Feature not active (no restrictions). This is the default.
2	Restrict stations with a COR assigned to LLC levels 2, 3, and 4.
3	Restrict stations with a COR assigned to LLC levels 3 and 4.
4	Restrict stations with a COR assigned to LLC level 4.

Precedence Call Timeout (sec)

Valid Entry	Usage
4 to 30	The number of seconds a precedence call remains in call waiting status before it is diverted. A busy user receives a precedence call waiting tone only if the incoming call cannot be connected and cannot preempt the user. The called party hears the tone every 10 seconds until answered or the administered time-out occurs. If ignored, the caller is diverted to an attendant or a call-forwarded station. The default is 30 seconds.

Preempt Emergency Call

If enabled, allows preemption of an Emergency 911 call made from a preemptable station by a higher precedence call.

Priority

The digit assignment for Priority precedence level calls. The default is 3.

Remote Attendant Route String

A user-defined telephone to which a precedence call can be routed when no console or night telephone is administered. Accepts from 1 to 24 digits.

Routine

The digit assignment for Routine precedence level calls. The default is 4.

WNDP Emergency 911 Route String

This route string is outpulsed when a user dials either 911 and waits for the interdigit timeout, or dials 911 followed by #. This dialing option only works when the WNDP Flash FAC is 91. Use a trunk access code (TAC), the AAR or the ARS access code, a WNDP access code, or an extension. For a WNDP access code, use the access code for the lowest precedence calling level in the system. Accepts from 1 to 24 digits.

Note:

An Emergency/911 call is a call that routes using the ARS table with the call type defined as either “alrt” or “emer”.

Comments on this document? infodev@avaya.com

Worldwide Numbering Dial Plan Active

Enables or disables the Worldwide Numbering Dial Plan. Disabled by default.

Chapter 91: Music Sources

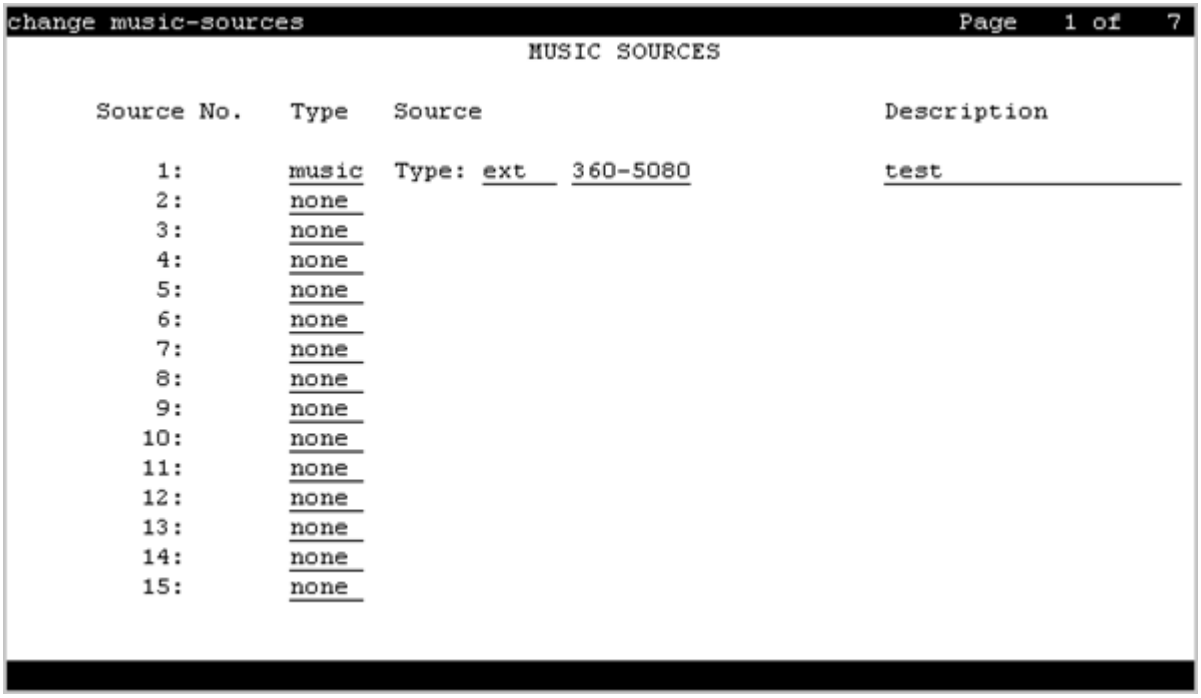
Use this screen to define music sources for Tenant Partitions. Each music source defined on this screen can be used by one or more Tenant Partitions. However, a partition can have only one music source.

Note:

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic Hold system, which does not require such a license, from Avaya Inc. or Avaya Business Partners.

Available only if **Tenant Partitioning** is enabled for the system.

Example command: `change music-sources`



```
change music-sources                                    Page   1 of   7
                                MUSIC SOURCES

    Source No.    Type   Source                              Description

        1:        music  Type: ext   360-5080                test
        2:        none
        3:        none
        4:        none
        5:        none
        6:        none
        7:        none
        8:        none
        9:        none
       10:        none
       11:        none
       12:        none
       13:        none
       14:        none
       15:        none
```

Related Links

Tenant Partitioning on page 806

Description

A description of the administered music source. Accepts up to 20 alphanumeric characters. Available only if music or tone is administered for treatment for hold.

Related Links

Type (Column) on page 598

Source

Available only if music is administered for treatment for hold.

Valid Entry	Usage
ext	Audio source extension for a single or group audio source.
group	A Music-on-Hold analog group number.
port	An analog or auxiliary trunk source location.

Related Links

Type (Column) on page 598

Source No

The number assigned to this music source.

Type (Column)

Valid Entry	Usage
music tone none	The type of treatment to be provided by the music source. Only one music source can use the tone value.

Type (field)

Available only if music is administered for treatment for hold.

Valid Entry	Usage
ext group port	The source for the music on hold. The source can be an announcement extension, an audio group, or a port on a VAL board. **Note:** A source identifier (extension number, audio group number, or port number) must be administered with the source type.

Related Links

Type (Column) on page 598

Chapter 92: Network Facilities

Used to administer new network-provided service or feature names and corresponding ISDN PRI (network specific facilities information element) encodings, for call-by-call trunk groups. Values for pre-defined facilities are displayed at the top of the screen and are display-only. User-defined facilities and services can be entered in the fields below.

When **Usage Allocation Enhancements** is enabled for the system, the system displays the Network Facilities screen. Additional user-defined entries can be administered on this screen.

Example command: `change isdn network-facilities`

Facility Coding

The ISDN-specified value for this service or feature.

Facility Type

The facility type. For types 2 and 3, **Usage Allocation Enhancements** must be enabled for the system.

Valid Entry	Usage
0 - feature	Predefined features.
1 - service	Predefined services.
2 - incoming	An incoming-type user-defined entry.
3 - outgoing	An outgoing-type user-defined entry.

Related Links

Usage Allocation Enhancements on page 806

Name

The name for the feature or service.

Chapter 93: Node Number Routing

Specifies the routing pattern associated with each node in a public or private network. Node Number Routing is a required capability for Extension Number Portability (ENP) and is associated with the Uniform Dial Plan (UDP).

Example command: `change node-routing n`, where *n* is the node number.

Node Number

The node number.

Partitioned Group Number

The partitioned group number associated with the node numbers being administered.

Route Pat

Valid Entry	Usage
1 to 2000 or blank	The routing pattern associated with the corresponding node number.

Chapter 94: Numbering-Private Format

Use this screen to specify the digits to be inserted in the calling number Information Element (IE), the connected number IE, and the QSIG party number for extensions in the Private Numbering Plan.

Communication Manager can generate private network numbers up to 15 digits long. Communication Manager can receive and display private network numbers up to 21 digits long.

Use the Numbering-Private Format screen to:

- Administer the extensions of all calls traversing the SIP trunks.
- Configure the private numbering table.
- Populate the calls to extensions terminating on a SIP-integrated Modular Messaging.

To change the value of the fields on the Numbering-Private Format screen, use the following command:

`change private-numbering` *n*, where *n* is the length of the extension.

The command **`change private-numbering`** *0* starts at the first entry.

Ext Code

In the case of a four-digit **Ext Len**, an **Ext Code** of 12 is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.

Valid Entry	Usage
0 to 13 blank	Accepts up to 13 digits depending on the administered extension length. When `0` alone is entered, the administered extension length must be 1 and the DDD number must be 10 digits.
attd	Generates a private calling number for a call from the attendant group.

Related Links

Ext Len on page 603

Ext Len

Valid Entry	Usage
0 to 13	The number of digits the extension can have.
blank	Corresponds to the extension lengths allowed by the dial plan.

Maximum Entries

The maximum number of private numbering entries that can be administered on the system.

Private Prefix

The number that is added to the beginning of the extension to form a Private Identification Number. The length of the prefix and the extension must at least equal the total length.

Total Administered

The number of private numbering entries currently administered on the system.

Total Len

Valid Entry	Usage
0 to 13	The total number of digits to send.

Trk Grp(s)

Communication Manager generates the station's identification number if **Ext Code** is administered, and this field is administered with the trunk group number carrying the call. Accepts one to seven digits. If blank, the identification numbers are independent on the trunk group carrying the call.

Related Links

Ext Code on page 602

Chapter 95: Numbering — Public/Unknown Format

Use this screen to specify the required digits for the Calling Number Information Element (IE) and the Connected Number Information Element (IE), in addition to the QSIG Party number, for any extension in Public and/or Unknown Number Plans. Use this screen for Automatic Route Selection (ARS) public trunks as well as SIP trunks. This screen supports the ISDN Call Identification Display feature. This feature provides a name and number display for display-equipped stations within an ISDN network. The system displays the name and number of the caller on the telephone display of the called party and displays the name and the number of the called party on the telephone display of the caller.

You must administer this screen if either **Send Calling Number** or **Send Connected Number** is specified, or **Supplementary Service Protocol** is b on the Trunk Group screen.

You must administer the following in the public numbering table on the Numbering-Public/Unknown Format screen:

- Extensions of all calls traversing SIP trunks
- Extensions terminating on SIP integrated Modular Messaging. However, do not configure a prefix for the SIP trunk group by which the Modular Messaging is integrated.

Note:

If an entry applies to a SIP connection to Avaya Aura® Session Manager, the resulting number must be a complete E.164 number. Communication Manager automatically inserts the plus (+) digit.

You can use the following command to modify the fields on this screen:

`change public-unknown-numbering` *n* , where *n* is the extension length from 0 to 13.

Related Links

Format on page 499
Send Calling Number on page 504
Send Connected Number on page 505
Supplementary Service Protocol on page 892

CPN Prefix

Use this field to enter the number that is added to the beginning of the extension to form a Calling Party Number (CPN) number.

You must enter only digits. You cannot enter leading spaces, or spaces in between the digits. You can enter up to 15 digits. Communication Manager formulates the CPN number in the following manner:

- If the length of **CPN Prefix** matches **Total CPN Length**, the extension number is not used to formulate the CPN number.
- If the number of digits in **CPN Prefix** plus the extension length exceeds the administered **Total CPN Length**, Communication Manager deletes excess leading digits of the extension when formulating the CPN number.
- If the number of **CPN Prefix** digits plus the extension length is less than the **Total CPN Length**, Communication Manager does not allow the entry.
- If the **Total CPN Length** is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.

Note:

If you leave this field blank, the extension is sent unchanged.

Ext Code

In the case of a four-digit **Ext Len**, an **Ext Code** of 12 is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.

Valid Entry	Usage
0 to 13 blank	Accepts up to 13 digits depending on the administered extension length. When 0 alone is entered, the administered extension length must be 1 and the DDD number must be 10 digits.
attd	Generates a private calling number for a call from the attendant group.

Related Links

Ext Len on page 603

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Ext Len

Valid Entry	Usage
0 to 13	The number of digits the extension can have.
blank	Corresponds to the extension lengths allowed by the dial plan.

Total CPN Len

Valid Entry	Usage
0 to 15	The number of digits the extension can have.
blank	Used when deleting an entry. This is the default.

Trk Grp(s)

Communication Manager generates the station's identification number if **Ext Code** is administered, and this field is administered with the trunk group number carrying the call. Accepts one to seven digits. If blank, the identification numbers are independent on the trunk group carrying the call.

Related Links

Ext Code on page 602

Chapter 96: Off-PBX Telephone Mobile Feature Extensions

Example command: `change off-pbx-telephone mobile-feature-ext`

Mobile Call (CTI) Extension

A CTI call to this Mobile Feature Extension (MCE) creates an OPTIM call under CTI influence. A call to the MCE triggers an OPTIM extend-call from a desk phone to its mapped cell phone number and to the destination. All calls made using the MCE appear to the destination as if they were dialed from the desk phone.

Chapter 97: Partition Routing Table

Identifies routing for partition groups associated with an ARS analysis entry.

Example command: `change partition-route-table n`, where *n* is the routing index.

PGN 1 (through PGN 8)

The routing for each partition group associated with each route index number.

Valid Entry	Usage
1 to 2000 for all templates except 254 for Avaya Aura® Main Embedded template.	The route pattern used to route the call.
r1 to r32	The remote home numbering plan area table used to route the call.
node	Node number routing.
deny	Blocks the call.

Chapter 98: Personal CO Line Group

Sets up a personal central office line group.

Example command: `add personal-CO-line` *n*, where *n* is the personal central office line group number.

Personal CO Line Group: page 1

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this personal central office line group generate call detail records. If **Record Outgoing Calls Only** is disabled for CDR, incoming calls on this personal central office line group also generate call detail records.
n	Calls over this personal central office line group do not generate call detail records.
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • **Abandoned calls**: The system creates a record with a condition code of “H” indicating the time until the call was abandoned. • **Answered calls**: The system creates a record with a condition code of “G” indicating the interval from start of ring to answer. • **Calls to busy stations**: The system creates a record with a condition code of “ I ” indicating a recorded interval of 0.

Related Links

Record Outgoing Calls Only on page 106

Coverage Path

Valid Entry	Usage
digits	The number of the call coverage path to use for incoming calls. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.
t1 to t999	The number of a time-of-day coverage table.
blank	No coverage path is assigned. Assigning a coverage path is optional.

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Related Links

Auto Answer on page 704

Group Name

A unique name that provides information about the personal CO line group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the personal CO line group rather than the group type (WATS).

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Display Character Set on page 787

Group Number

The personal CO line group number.

Group Type

The type of personal central office line group. The fields that are displayed and available might change according to the personal central office line group type selected.

Valid Entry	Usage
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can route to certain areas (“WATS band”) for a flat monthly charge. With incoming WATS trunks, toll-free calling can be used to call customers and employees.

Outgoing Display

With Outgoing Display, telephones can show the name and number of the personal central office line group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the personal central office line group name and number.
n	Displays the digits the caller dials.

Security Code

The station security code that users must dial to retrieve voice messages and to use the Demand Print Message feature. Accepts from three to eight digits.

TAC

The trunk access code (TAC) that must be dialed to access the personal central office line group. A different TAC must be assigned to each personal central office line group. CDR reports use the TAC to identify each personal central office line group. The characters “*” and “# ” can be used as the first character in a TAC. Accepts a one- to four-digit number.

TRUNK PARAMETERS

Analog Loss Group

Determines which administered two-party row in the loss plan applies to this personal central office line group if the call is carried over an analog signaling port in the personal central office line group.

Valid Entry	Usage
1 to 17	The index into the loss plan and tone plan. If values are administered other than in between 6 and 10 or 15 and 17, a warning message displays stating that the loss group might not be appropriate for this personal central office line group.

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Note:

This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers.

Related Links

Receive Answer Supervision on page 617
Administer Timers on page 490

Call Still Held

If enabled, the system prevents glare by extending the Incoming Glare Guard timer and delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call. This field is used when the receiving end media server or switch initiates the disconnection of incoming calls. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the **Country** code is 2.

Available only for co or fx personal central office line groups.

Related Links

Group Type on page 611

Charge Conversion

Available only for outgoing or two-way CO, FX, and WATS trunk groups.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related Links

Direction on page 481
Trunk Direction on page 617

Charge Type

Text string used to describe charges related to a telephone call. The system displays these words or characters on telephone screen after the charge amount. Typically uses either the currency symbol or the charge type, but not both. Accepts up to seven characters. Embedded spaces count as characters.

Available only for outgoing or two-way CO, FX, and WATS personal central office line groups.

Related Links

Direction on page 481

Country

The country code that corresponds to the protocol used by the local telephone company central office (CO) where the personal central office line group terminates.

Available only for personal central office line groups that connect Communication Manager to a CO in the public network — CO, FX, and WATS personal central office line groups.

Valid Entry	Usage
1 to 25, except for 19	For a list of country codes, see the *Country code table*.

Related Links

Country options table on page 783
Trunk Gain on page 617
Trunk Termination on page 618

Currency Symbol

The telephone displays the currency symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, FX, and WATS personal central office line groups.

Related Links

Trunk Direction on page 617
Direction on page 481

Decimal Point

The appropriate representation for a decimal point as seen on the telephone displays. Available only with outgoing or two-way CO, FX, and WATS personal central office line groups. Also available for isdn personal central office line groups, if the **Charge Advice** field is set to a value other than none.

Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG personal central office line group, unlike other personal central office line groups, the **Decimal Point** field is independent of whether the calling screen displays a decimal point. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related Links

Charge Advice on page 480
Direction on page 481
Trunk Direction on page 617

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this personal central office line group if the call is carried over a digital signaling port in the personal central office line group. If values other than 18 or between 11 and 15 are administered, a warning message displays stating that the loss group is inappropriate for this personal central office line group.

Disconnect Supervision-In

Indicates whether Communication Manager receives disconnect supervision for incoming calls over this personal central office line group.

Available only for incoming or two-way personal central office line groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end server or switch sends a release signal when the calling party releases an incoming call, and the far-end server or switch is responsible for releasing the trunk. Enhances Network Call Redirection.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on incoming calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

In general, U.S. local telephone company central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with the network services provider.

Related Links

Direction on page 481
Trunk Direction on page 617

Outgoing Dial Type

Sets the method used to transmit digits for an outgoing call. Usually, this method should match what the local telephone company central office provides.

Available for CO, FX, and WATS personal central office line groups. This field is also available for the trunks with the **group type** as Tie and when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Uses Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. With this, the personal central office line group can support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
rotary	With this, only the dial pulse addressing method can be used by non-touch tone telephones. For example, this value is appropriate for an internal full touch tone system and for a connection to a local telephone company central office that only supports rotary dialing.
automatic	For tie trunks if the **Trunk Signaling Type** is not administered. This provides “cut-through” operation to outgoing callers who dial a trunk access code, connecting them directly to local telephone company central office dial tone and bypassing any toll restrictions administered on Communication Manager.
mf	Use this option if the **Trunk Signaling Type** field is blank. The system displays this field only if you set the **Multifrequency Signaling** field on the System Parameters Customer-Options screen to y. You cannot use this option if the **Used for DCS** field is y. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
r1mf	Enter r1mf for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter r1mf to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. Group type field must be set to co.
mf2/6	Use this entry for signaling in Spain. This signaling type provides normal public network service and a virtual private network digital trunk service called IBERCOM. Use this entry only if the **Trunk Signaling Type** field is blank. The system displays this entry only if you enable **Multifrequency Signaling** for the system. The system does not display this entry if the trunk is used for DCS.

Related Links

Group Type on page 482
Multifrequency Signaling on page 802
Trunk Signaling Type on page 875

Prefix-1

If enabled, the prefix “1” is added to the beginning of the digit string for outgoing calls. Use this field for outgoing and two-way personal central office line groups handling long distance service. Disable for personal central office line groups in AAR or ARS route patterns.

Available only for CO and FX personal central office line groups.

Receive Answer Supervision

If enabled, the Communication Manager provides answer supervision for a trunk group. For Outbound Call Management applications, use for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Related Links

Answer Supervision Timeout on page 491
Administer Timers on page 490

Trunk Direction

Important:

This setting must match the provider's settings.

The direction of the traffic on this personal central office line group.

Valid Entry	Usage
incoming	Traffic on this personal central office line group is incoming.
outgoing	Traffic on this personal central office line group is outgoing.
two-way	Traffic on this personal central office line group is incoming and outgoing. Required for **Network Call Redirection**.

Related Links

Charge Conversion on page 497

Trunk Gain

Specifies the amplification applied to the trunks in this group. With the values administered for **Trunk Termination** and **Country** code, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO circuit packs are set automatically to high.

Valid Entry	Usage
high	Used if users complain of low volume.
low	Used if users complain of squeal or feedback.

Related Links

Trunk Termination on page 618
Country on page 613

Trunk Port

The port to which the trunk is connected.

Trunk Type

Controls the seizure and start-dial signaling used on this personal central office line group. Entries in this field vary according to the function of the personal central office line group and must match the corresponding setting on the far-end server or switch.

Available only for CO, FX, and WATS personal central office line groups.

Valid Entry	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible. Ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, loop-start signaling is used only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto auto/delay auto/immed auto/wink	The term before the slash tells Communication Manager how and when it receives incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming central office trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it is ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch. • wink — The sending server running Communication Manager does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.
2-wire-ac 2-wire-dc 3-wire	These entries are used with local telephone company central office (CO) trunks in Russia. The specific CO should match one of these values. Available only if the **Country** code is 15 and the CO trunks use ports on a TN2199 circuit board.

Related Links

Country on page 613

Trunk Termination

Adjusts the impedance of the personal central office line group for optimal transmission quality.

Valid Entry	Usage
600ohm	The distance to the local telephone company central office (CO) or to the server at the other end of the trunk is less than 3,000 feet.
rc	The distance to the CO or to the server at the other end of the trunk is more than 3,000 feet.

Personal CO Line Group: page 2

Ext

The extension of telephones that have a **CO Line** button.

Name

The name assigned to telephones that have a **CO Line** button.

Chapter 99: Pickup Group

Implements call pickup groups. A pickup group is a group of users authorized to answer calls to a telephone extension within that group of users. A telephone extension can belong to only one pickup group, and a pickup group can have up to 50 extensions.

Example command: `add pickup-group n`, where *n* is the pickup group number.

GROUP MEMBER ASSIGNMENTS

Ext

The extension of the pickup group number. A VDN cannot be assigned to a Call Pickup group.

Name

The name assigned to the pickup group number extension.

Extended Group Number

Available only if the **Extended Group Call Pickup** is administered as flexible.

Valid Entry	Usage
1 to 100 blank	The Extended Group number. The extended group is a collection of pickup groups that can answer calls to a telephone extension within that other pickup group.

Related Links

Extended Group Call Pickup on page 282

Group Number

The pickup group number.

Chapter 100: Policy Routing Table

With Policy Routing Table, you can distribute calls among a set of call centers based on specified percent allocation. Various types of incoming calls that arrive at a particular VDN can be directed to a Policy Routing Table (PRT) instead of to a vector. The PRT then distributes the calls to the administered Route-to VDNs based on the specified percent allocation targets. Use this screen to implement and monitor percentage allocation routing by assigning destination routes and target percentages.

Example command: `change policy-routing-table n`, where *n* is the policy routing table number.

Actual %

The actual percent of total calls routed to a VDN. Calculated to six decimal places, but only the first decimal place is displayed.

Call Counts

The current number of calls routed to a VDN.

Index

The sequential number of the row in a Policy Routing Table.

Name

The name of the Policy Routing Table (PRT). Accepts a string of up to 15 alphanumeric characters.

Number

The number of the Policy Routing Table (PRT).

Period

The period for resetting the call counts and actual percentages.

Valid Entry	Usage
100_count	Resets the call counts (and displayed %) when total calls for the PRT reach 100. At this point, the total calls match the target routing pattern percentages. This ensures that the routing points have equal distribution of calls all the time. This is the default.
max_count	Call counts are maintained until calls delivered to at least one of the VDNs exceed 65,400. At this point, calls continue to be distributed over the VDNs but the call counts are reset when the actual percentages equal the targets for all of the VDNs at the same time.
Half-hour	Resets the call counts at the top of the hour and at the 30–minute point.
hour	Resets the call counts at the top of the hour.
daily	Resets the call counts at midnight, every night.
weekly	Resets the call counts at midnight on Saturday.

Route-to VDN

VDN extension to which calls are to be routed. Accepts up to 15 extensions containing 1 to 13 digits.

Target %

Valid Entry	Usage
0 to 100	The target percent of total calls to be routed to a VDN. Use whole numbers only, no decimal fractions.

Totals

Displays totals for **Target %** and **Call Counts** for all the assigned VDNs in the PRT. The total for **Target %** is always 100 for form submittal.

Type

The type of algorithm the Policy Routing Table (PRT) supports.

VDN Name

The name of a VDN, if a name has been assigned previously.

Chapter 101: Precedence Routing Digit Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern of an outgoing Multiple Level Precedence and Preemption (MLPP) call.

Example command: `change precedence-routing analysis n`, where *n* is the precedence-routing digit analysis table number.

Destination Code Control

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Use Destination Code Control (DCC) to restrict outbound calls to a specific called number or a range of numbers to keep the trunk free for important calls.

DCC limits outbound calls from the outgoing budget (IPBo).

To enable the DCC feature you must first activate DCC by using FAC.

Valid Entry	Usage
y	DCC is activated.
n	DCC is not activated. The default value is n.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura*® *Communication Manager System Capacities Table*, 03-300511.

Percentage IPBO Blocked

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

With **Percentage IPBO Blocked**, Communication Manager supports blocking a percentage of calls to a destination for the DCC feature. You can enter a value from 1% to 100%. The default value is `100%`.

Preempt Method

The preemption method used by the server running Communication Manager for this dialed string.

Valid Entry	Usage
group	The system checks the first trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the system connects the call. This is the default.
route	The system checks each trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the call is connected.

Chapter 102: Precedence Routing Digit Conversion Table

Assigns the Precedence Routing digit conversion. Digit conversion takes digits dialed on incoming calls and converts the digits to local telephone numbers, usually extension numbers.

Example command: `change precedence-routing digit-conversion` *n*, where *n* is the precedence routing digit conversion table number.

Conv

Use this field to enable additional digit conversion.

Del

Use this field to enter the number of digits that the system must delete from the starting of the dialed string.

Matching Pattern

Use this field to specify the pattern that the system must use to match the dialed number. The pattern can be up to 18 characters long and consist of digits and the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Net

Valid Entry	Usage
ext	Communication Manager routes the call as an extension using the Dial Plan Analysis and Uniform Dial Plan forms, as appropriate.
pre	Precedence routing. Uses the Precedence Analysis Tables to route the call.

Replacement String

Use this field to specify the digit string that must replace the deleted portion of the dialed number.

Valid entry	Usage
blank	To delete the digits without replacement.
0 to 9, *	To type the digit string. The digit string can have up to 18 characters.
#	To indicate the end of dialing.

Chapter 103: PRI Endpoint

Administers PRI Endpoints for the Wideband Switching feature.

Note:

A PRI Endpoint with a width greater than 1 can be administered only if Wideband Switching has been enabled for the system.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system.

A PRI Endpoint is defined as 1 to 31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

Example command: `add pri-endpoint n`, where *n* is the extension number.

Related Links

Wideband Switching on page 807

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) used to determine calling and called party privileges.

COS

Valid Entry	Usage
0 to 15	The Class of Service (COS) used to determine the features that can be activated by, or on behalf of, the endpoint.

Extension

The extension number used to access the PRI endpoint.

Maintenance Tests

If enabled, runs hourly maintenance tests on this PRI Endpoint.

Name

The name of the endpoint. Accepts up to 27 alphanumeric characters.

Originating Auto Restoration

If enabled, automatically restores calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid Entry	Usage
digits or blank	The D-channel or D-channel pair that provides the signaling information for the set of B-channels that make up the PRI Endpoint. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Simultaneous Calls

If enabled, specifies that multiple simultaneous calls can be placed to/from the PRI Endpoint.

(Starting) Port

The seven-character starting port of the PRI Endpoint.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.

Table continues…

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Valid Entry	Usage
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Width

Valid Entry	Usage
1 to 31	The number of adjacent DS0 ports beginning with the administered Starting Port, that make up the PRI Endpoint. This field cannot be blank. A width of 6 defines a PRI Endpoint that can support data rates up to 384 Kbps.

Related Links

(Starting) Port on page 631

WIDEBAND SUPPORT OPTIONS

Contiguous

Available only if “N by DS-zero” (NXDS0) multi-rate service is enabled.

Valid Entry	Usage
y	Specifies the “floating” scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk). Unavailable with H0 ISDN information transfer rate.
n	Specifies the “flexible” scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points

Related Links

NxDS0 on page 519
NXDS0 on page 633

H0

If enabled, specifies the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

If enabled, specifies the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

If enabled, specifies the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

If enabled, specifies the NXDS0 multi-rate service.

Chapter 104: Processor Channel Assignment

Assigns each local processor channel to an interface link channel, and defines the information associated with each processor channel on an Ethernet link.

Note:

You cannot remove a service from this screen if that service has overrides defined on the Survivable Processor screen.

Example command: `change communication-interface processor-channels`

Appl

Specifies the server application type or adjunct connection used on this channel.

Valid Entry	Usage
audix	Voice Messaging.
ccr	Contact Center Reporting, now known as Avaya IQ.
dcs	Distributed Communication System.
fp-mwi	ISDN Feature Plus Message Waiting Indication. This channel passes message waiting light information for subscribers on the messaging system, from a messaging adjunct on a main switch for a telephone on a satellite switch. The terminating location (far end) of this channel must be a Communication Manager system compatible with ISDN Feature Plus proprietary protocol.
gateway	Supports an X.25 connected AUDIX connected to an ISDN DCS network.
gateway-tcp	Supports a TCP-connected voice messaging system connected to an ISDN DCS network.
mis	Management Information System, otherwise known as the Call Management System.
qsig-mwi	QSIG Message Waiting Indication. Used with a QSIG-based interface to a messaging system, this channel passes message waiting light information for subscribers on the messaging system.

Destination Node

Identifies the server or adjunct at the far end of this link. The destination node can be an adjunct name, server name, far end IP address, or node name for services local to the Avaya server. For ppp connections, match the Destination Node Name administered for the ppp data module.

Destination Port

Valid Entry	Usage
5000 to 64500	The number of the destination port.
0	Any port can be used.

Enable

Enables or disables this processor channel on the main server.

Gtwy to

A number that identifies the processor channel to which the specified processor channel is serving as a gateway.

Interface Channel

The channel number or the TCP/IP listen port channel to carry this processor (virtual) channel.

Valid Entry	Usage
5000 to 64500	For ethernet or ppp. For TCP/IP, interface channel numbers are in the range 5000 to 64500. The value 5001 is recommended for CMS, and 5003 is recommended for DCS.
0	Any port can be used.

Interface Link

Valid Entry	Usage
1 to 254	The physical link carrying this processor (virtual) channel.
pv4 or pv6	Communication Manager's Processor Ethernet interface used for adjunct connectivity.
blank	Not administered.

Mach ID

Valid Entry	Usage
1 to 63 for MWI 1 to 63 for DCS 1 to 99 for voice messaging blank	The destination server ID defined on the dial plan of the destination server.

Related Links

Mach ID on page 694

Mode

Valid Entry	Usage
c(lient) s(erver) blank	Indicates whether the IP session is passive (client) or active (server). This field must be blank if the interface link is procr-intf. This field cannot be blank if the type of interface link is ethernet or ppp.

Proc Chan

The number assigned to each processor channel.

Session - Local/Remote

Valid Entry	Usage
1 to 384 blank	The Local and Remote Session numbers. For each connection, the Local Session number on the Avaya server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections. Local and Remote Session numbers must be consistent between endpoints.

Chapter 105: QSIG to DCS TSC Gateway

This screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

Available only if **Interworking with DCS** is enabled for the system.

Example command: `change isdn qsig-dcs-tsc-gateway`

Related Links

Interworking with DCS on page 814

Sig Grp

Valid Entry	Usage
digits	The assigned signaling group number. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Subscriber Number

A subscriber number up to 20 characters in length. Accepts *, x, and X as wildcard characters.

Note:

If the Subscriber Number is local, the form does not send any DCS messages over the listed Sig Grp/TSC Index. Therefore, you can put a block entry like '2xxx' with the understanding that if extension 2001 is local, no DCS message will be sent.

TSC Index

The TSC Index for each machine ID.

Valid Entry	Usage
1 to 64	The assigned signaling group number for the qsig-mwi application type.

Chapter 106: Reason Code Names

Assigns names to reason codes. Each reason code can have a different name for Aux Work and for Logout. The system displays the screen only when you set the **Two-Digit Aux Work Reason Codes** to y.

Note:

Logout reason codes can only be in the range of 0 to 9, even if **Two-Digit Aux Work Reason Codes** is active.

Example command: `change reason-code-names`

Aux Work

The name associated with a reason code when the agent uses the reason code to enter Aux Work mode. Accepts up to 16 alphanumeric characters.

Default Reason Code

The default reason code names. A separate name can be administered for the Aux Work Reason Code of 0 and for the Logout Reason Code of 0. If an agent changes to Aux Work mode and the Aux Work Reason Code Type is set to none, the agent is put into Aux Work mode with the default Aux Work reason code, even if a different reason code is administered for the **Aux** button. If an agent logs out when the Logout Reason Code Type is set to none, the agent is logged out with the default Logout reason code.

Accepts up to 16 alphanumeric characters.

Interruptible

Specifies whether or not each reason code is interruptible.

Note:

The Default Reason Code, Auto-answer IP Failure Aux Work Reason Code, and Maximum Agent Occupancy Aux Work Reason Code cannot be made interruptible.

Logout

The name associated with a reason code when the agent uses the reason code to log out. Accepts up to 16 alphanumeric characters.

Chapter 107: Remote Access

Implements the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Security alert:

The Remote Access feature, when properly administered, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya will not be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Example command: `change remote-access`

Authorization Code Required

If enabled, requires an authorization code be dialed by Remote Access users to access the system's Remote Access facilities. The use of an authorization code in conjunction with a barrier code increases the security of the Remote Access feature.

Barrier Code

The number users must dial to use Remote Access. Can be used with an authorization code. Accepts a four- to seven-digit number in any combination of digits. Must conform to the administered **Barrier Code Length**. The none value is required if a **Barrier Code Length** is not administered.

Related Links

Barrier Code Length on page 643

Barrier Code Length

Valid Entry	Usage
4 to 7 blank	The length of the barrier code. A barrier code length of 7 provides maximum security.

Calls Used

The number of calls placed using the corresponding barrier code. A usage that exceeds the expected rate indicates improper use.

COR

Valid Entry	Usage
0 to 995	The number of the class of restriction (COR) associated with the barrier code. To provide maximum security, assign the most restrictive COR that provides only the level of service required.

COS

Valid Entry	Usage
0 to 15	The number of the class of service (COS) associated with the barrier code. To provide maximum security, assign the most restrictive COS that provides only the level of service required.

Disable Following a Security Violation

Available only if SVN Authorization Code Violation Notification is enabled for the system.

Valid Entry	Usage
y	Disables the Remote Access feature following detection of a Remote Access security violation. The system administrator can re-enable Remote Access using the `enable remote-access` command.

Table continues…

Valid Entry	Usage
n	Enables the Remote Access feature following detection of a Remote Access security violation.

Related Links

SVN Authorization Code Violation Notification Enabled on page 662

Expiration Date

The date the barrier code expires. Assign an expiration date based on the expected length of time the barrier code is needed. For example, if the barrier code is expected to be used for a two-week period, assign a date two weeks from the current date. If both an **Expiration Date** and **No. of Calls** are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

No. of Calls

Valid Entry	Usage
1 to 9999 blank	The number of Remote Access calls that can be placed using the associated barrier code. If both an **Expiration Date** and **No. of Calls** are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Related Links

Expiration Date on page 644

Permanently Disable

If enabled, permanently blocks remote access to the administration interface. Reactivation of remote access to the interface requires the intervention of Avaya Services.

Remote Access Dial Tone

Enables or disables the user hearing dial tone as a prompt for entering the authorization code. Disabling this feature provides maximum security. Available only if Authorization Codes are required for Remote Access.

Related Links

Authorization Code Required on page 642

Remote Access Extension

The extension assigned to handle Remote Access calls. The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered as the trunk group's incoming destination.

A Vector Directory Number (VDN) extension cannot be used as the remote access extension.

Can be blank if no barrier codes are administered.

Related Links

Incoming Destination on page 869

TN

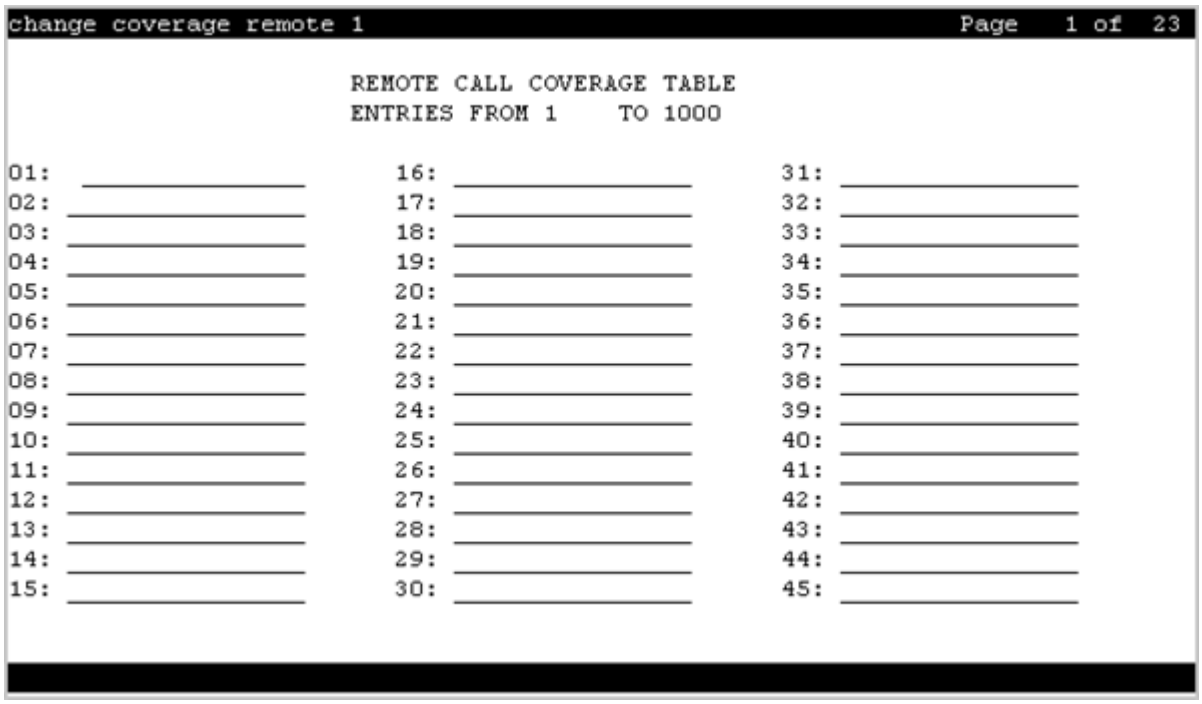
Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 108: Remote Call Coverage Table

Provides automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers can be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits or blank, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code.

Example command: `change coverage remote n`, where *n* is the remote coverage number.



01–1000

The identifier for the destination coverage point. Accepts up to 16 digits. Two places are used for L, D, ',', and %.

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Valid Entry	Usage
*	DTMF digit asterisk
#	DTMF digit pound
L	Uses a coverage point only when in Survivable Remote or Survivable Core Server mode
D	Represents the called extension digits
’	Pauses for 1.5 seconds
%	Remaining digits are for end-to end signaling
blank	Not administered

Chapter 109: Remote Office

Supports the Remote Office feature, an arrangement whereby a user can set up a remote office without having an on-premises physical desk-set. An R300 is issued to connect remote DCP and analog telephones, IP telephones, and H.323 trunks to the Communication Manager server using IP.

Example command: `change remote-office n`, where *n* is the assigned remote office number.

Location

Valid Entry	Usage
1 to 2000	(Depending on your server configuration, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.) Assigns the location number to the remote office comprised of the associated time zone and the appropriate numbering plan. See the Location sections in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205, for the other ways to assign a location, and for a list of features that use location.
blank	Obtains the location from the cabinet containing the CLAN or the gateway that the endpoint registered with. By default, the value is blank.

Network Region

Valid Entry	Usage
1 to 2000 blank	The network region assigned to all stations supported at this remote office.

Node Name

The node name of the remote office.

Site Data

Any required site information. Accepts up to 30 alphanumeric characters.

Chapter 110: RHNPA Table

Defines route patterns for specific three-digit codes, usually direct distance dialing (DDD) prefix numbers.

Example command: `change rhnpa n`, where *n* is the prefix number.

CODES

The 100-block of codes being administered.

Code-Pattern Choice Assignments

Valid Entry	Usage
1 to 24	A pattern choice number associated with each office code.

Pattern Choices

Valid Entry	Usage
1 to 999 1 to 2000 blank	The maximum number of route patterns that you can configure depends on your system configuration. The route pattern number associated with each code. If you use one pattern for most of the codes, assign that pattern to choice 1.

RHNPA TABLE

The RHNPA table number.

Chapter 111: Route Pattern

Use this screen to define route patterns. Each route pattern contains a list of trunk groups that Communication Manager uses to route a call. The maximum number of route patterns and trunk groups allowed depends on the system configuration and the memory available in the system.

Use this screen to convert the dialed number into a format that the network uses for routing the call. For example, you can:

- Insert or delete digits when Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) calls route over different trunk groups.
- Convert an AAR number into an international number.
- Insert an area code into an AAR number to convert an on-network number to a public network number.
- Insert the dial access code for an alternative carrier. The system uses this alternative carrier when a call directly accesses a local telephone company central office (CO) and if the long distance carrier that the CO provides is unavailable.

To modify the value of the fields on the Route Pattern screen, use the following command:

`change route-pattern` *n*, where *n* is the route pattern number.

Band

A number that represents the OUTWATS band number (US only).

Available only if **Services/Features** is administered as outwats-bnd and **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system. **Band** is required by Call-by-Call Service Selection.

Related Links

ISDN-BRI Trunks on page 801
ISDN-PRI on page 801
Service/Feature on page 659

BCC Value

Identifies the type of call appropriate for a trunk group. Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
y/n	If enabled, (0, 1, 2, 3, 4, or W) indicates the BCC value is valid for the routing preference. A trunk group preference can have more than one BCC.

BCC values:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
M	Multimedia call
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

Related Links

ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

BCIE (Bearer Capability Information Element)

Determines how to create the ITC codepoint in the setup message. Applies to ISDN trunks. Available only if the Information Transfer Capability (ITC) is administered as both.

Valid Entry	Usage
ept	endpoint
unr	unrestricted

Related Links

ITC (Information Transfer Capability) on page 654

CA-TSC Request

For the DCS+ features, Communication Manager sends a Call-Associated Temporary Signaling Connection (CA-TSC) request to send messages other than normal call control messages.

Valid entry	Usage
as-needed	Communication Manager sends the **CA-TSC** request only when needed.

Table continues…

Valid entry	Usage
at-setup	Communication Manager sends the **CA-TSC** request for every B-channel call, whether or not it is needed.
none	Communication Manager does not send the **CA-TSC** request Use this entry to allow Communication Manager to send Non-Call-Associated Temporary Signaling Connection (NCA-TSC) requests.

DCS/QSIG Intw

Enables or disables DCS/QSIG Voice Mail Interworking. Available only if **Interworking with DCS** is enabled for the system.

Related Links

Interworking with DCS on page 814

FRL

Valid Entry	Usage
0 to 7	The Facility Restriction Level (FRL) associated with the group routing preference. 0 is the least restrictive, and 7 is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk group.

Security alert:

For system security reasons, use the most restrictive FRL possible.

Grp No

Valid Entry	Usage
1 to 2000	The trunk group number associated with the routing preference.

Hop Lmt

The number of hops for each routing preference. Communication Manager blocks a hop equal to or greater than the number entered.

Valid Entry	Usage
blank	There is no limit to the number of hops for this preference.
1 to 9	Valid for limiting the number of hops if using the tandem hop feature.
1 to 32	Valid for limiting the number of hops if using the transit feature.

Inserted Digits

The digits to insert for routing. Communication Manager can send up to 52 digits. This includes up to 36 digits entered here plus up to 18 digits originally dialed. Special symbols count as two digits each.

Valid Entry	Usage
*	When * is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the local telephone company central office (CO) in place of the *.
#	When # is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the CO in place of the #.
, (comma)	Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone depending on system-wide feature administration. The length of waiting time is controlled by the **Off-premises Tone Detect Timeout Interval** field on the Feature-related system parameters screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI that is used for Russian pulse trunks.
p	The associated trunk group must be of type sip. The single digit p is used for fully qualified E.164 numbers. Communication Manager inserts a plus (+) digit to the digit string.

Related Links

Off-Premises Tone Detect Timeout Interval (seconds) on page 272
Outpulse Without Tone on page 310

ITC (Information Transfer Capability)

Identifies the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be both or unre for a **BCC Value** of W.

Valid Entry	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ricted)	Calls from restricted endpoints can access the route pattern.
unre(stricted)	Calls from unrestricted endpoints can access the route pattern.

Related Links

BCC Value on page 651

IXC

Identifies the carrier, such as AT&T, used for calls that route using an Inter-Exchange Carrier (IXC), and for Call Detail Recording (CDR).

Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
Valid carrier code	Identifies the carrier for IXC calls.
user	For presubscribed carrier. Used when an IXC is not specified.
none	Must be none for non-ISDN trunk groups and for Telcordia Technologies NI-2 Operator Service Access. If it is necessary to send an IXC code for a non-ISDN trunk group, the IXC code can be entered as **Inserted Digits**.

Related Links

Inserted Digits on page 654
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

LAR

The routing-preference for Look Ahead Routing.

Valid Entry	Usage
next	Go to the next routing preference and attempt the call again.
rehu	Re-hunt within the current routing preference for another trunk to attempt the call again.
none	Look Ahead Routing is disabled for the preference.

No. Del. Digits

Modifies the dialed number so an AAR or ARS call routes over different trunk groups that terminate in servers or switches with different dial plans.

Valid Entry	Usage
0 to 28 blank	Inter-Exchange Carrier (IXC) digits are deleted if the field is administered to any non-blank value. If the field is blank, no IXC digits are deleted. Communication Manager administers the IXC digits on the Inter-Exchange Carrier (IXC) Codes screen. Use for calls that route: • To or through a remote server. • Over tie trunks to a private network server. • Over local telephone company central office (CO) trunks to the serving CO.

No. Dgts Subaddress

Available only if **ISDN Feature Plus** is enabled for the system.

Valid Entry	Usage
1 to 5 blank	The number of dialed digits to send in the calling party subaddress IE. Allows a caller to reach a number where the media server's digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant.

Related Links

ISDN Feature Plus on page 801

NPA

The three-digit Numbering Plan Area (NPA) or area code for the terminating endpoint of the trunk group. Not required for AAR.

The local telephone company can verify this number. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.

Leave blank for AAR calls and for tie trunks.

Numbering Format

Use this field to specify the numbering format of calls that Communication Manager routes over trunk group of the preferred routing pattern. This field applies to ISDN and SIP trunk groups.

Valid entry	Numbering Plan Identifier	Type of Numbering
blank	Derived from the value of the **Call Type** field on the ARS Analysis screen or the AAR Analysis screen.	Derived from the value of the **Call Type** field on the ARS Analysis screen or the AAR Analysis screen.
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

Note:

To gain access to Telcordia Technologies NI-2 Operator Service Access, type `unk-unk` in the **Inserted Digits** field.

For Network Call Redirection or Transfer, type `lev0-pvt` in the **Numbering Format** field.

Pattern Name

An alphanumeric name for the route pattern.

Pattern Number

The route pattern number.

Prefix Mark

Sets the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when

call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

Not required for AAR. ARS requires a number from 0 to 4 or blank.

For a WATS trunk, the **Prefix Mark** is the same as the local telephone company central office (CO) trunk.

Valid Entry	Usage
0	• Suppresses a user-dialed prefix digit 1 for 10-digit FNPA calls. • Leaves a user-dialed prefix digit 1 for 7-digit HNPA calls. • Leaves a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Should not be used in those areas where all long-distance calls must be dialed as 1+10 digits. Check with the local network provider.
1	Sends a 1 on 10-digit calls, but not on 7-digit calls. Used for HNPA calls that require a 1 to indicate long-distance calls.
2	Sends a 1 on all 10-digit and 7-digit long-distance calls. Refers to a Toll Table to define long-distance codes.
3	Sends a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long-distance calls are 10-digit calls. The NPA is inserted when a user dials a prefix digit 1 plus 7 digits. Refers to a Toll Table to define long-distance codes.
4	Always suppress a user-dialed prefix digit 1. Used, for example, when ISDN calls route to a server that rejects calls with a prefix digit 1.
blank	For tie trunks.

SCCAN

If enabled, indicates that the route pattern supports incoming SCCAN calls. Available only if Enhanced EC500 is enabled for the system.

Related Links

Enhanced EC500 on page 799

Secure SIP

If enabled, specifies using the SIP or SIPS prefix when the call is routed to a SIP trunk preference. If SIP trunks are not specified, the call is routed over whatever trunk is specified.

Service/Feature

An identifier of the Service/Feature carried by the information element (IE) in a call in this route preference. Required by Call-by-Call Service Selection, and Network Call Redirection Transfer. Accepts up to 15 characters.

Note:

User-defined service types for network facilities can also be used. Any user-defined **Facility Type** of 0 (feature), 1 (service), or 3 (outgoing) is allowed.

Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry		
accunet	multiquest	`sdn` (allows Network Call Redirection/Transfer)
i800	operator	sub-operator
lds	oper-meg (operator and megacon)	sub-op-meg (sub-operator and megacom)
mega800	oper-sdn (operator and sdn)	sub-op-sdn (sub-operator and sdn)
megacom	outwats-bnd	wats-max-bnd

Related Links

Facility Type on page 600
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Toll List

Valid Entry	Usage
1 to 32 blank	The number of the ARS Toll Table associated with the terminating NPA of the trunk group. Required for **Prefix Mark** 2 or 3. Not required for AAR.

Related Links

Prefix Mark on page 657

TSC

Use the **Temporary Signaling Connection (TSC)** field to support feature transparency on DCS+ calls and to support QSIG call completion. You must enable **TSC** to allow Communication Manager to send Call-Associated TSC (CA-TSC) and Non-Call-Associated TSC (NCA-TSC) requests for a routing preference.

Chapter 112: Security related system parameters

Determines when Avaya Communication Manager reports a security violation. Many of the fields on this screen repeat for each type of security violation. They are explained once here, but the usage is the same for all.

Example command: `change system-parameters security`

Security related system parameters: page 1

Hide Post-Dialing DTMF On List Trace

Enable this field to display asterisk (*) characters in place of digits in the output of the list trace command. The default value of this field is n.

SECURITY VIOLATION NOTIFICATION PARAMETERS

Announcement Extension

The announcement extension where the Security Violation Notification (SVN) announcement resides. The server running Communication Manager calls the referral destination, then plays this announcement upon answer.

Originating Extension

The extension that initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination. If notification for more than one type of security violation is established, a different extension must be assigned to each one. When Communication Manager generates a referral call, this extension and the type of violation appear on the display at the referral destination.

Referral Destination

The extension that receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless it is an Announcement Extension. The extension can be the telephone, attendant console, or vector directory number (VDN) that receives the referral call for each type of violation. This can be the same extension for all type of violations.

The **Announcement Extension** field is used for a VDN. Call Vectoring Time-of-Day routing is used to route the referral call to different destinations based on the time of day or the day of the week.

Related Links

Announcement Extension on page 661

SVN Authorization Code Violation Notification Enabled

Enables or disables Authorization Code Violation Security Notification. Use with **SVN Remote Access Violation Notification Enabled** to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. The system cannot disable remote access following a security violation unless this field has been enabled.

Related Links

SVN Remote Access Violation Notification Enabled on page 662

SVN Login (Violation Notification, Remote Access, Authorization Code) Enabled

Enables or disables login violation notification. If enabled, Communication Manager sends a notification when a login violation occurs.

SVN Remote Access Violation Notification Enabled

Enables or disables Remote Access Violation Notification. Use with **SVN Authorization Code Violation Notification Enabled** to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. The system cannot disable remote access following a security violation unless this field is enabled.

Related Links

SVN Authorization Code Violation Notification Enabled on page 662

Time Interval

Valid Entry	Usage
0:01 to 7:59	This time range, in conjunction with **Login Threshold**, determines if a security violation has occurred. The range for the time interval is one minute to eight hours, entered in the screen x:xx. For example, one minute is entered as 0:01 and seven and one-half hours is entered as 7:30.

Security related system parameters: page 2

ACCESS SECURITY GATEWAY PARAMETERS

These fields are available only if **Access Security Gateway (ASG)** is enabled for the system.

Related Links

Access Security Gateway (ASG) on page 794

EPN

Indicates whether or not any entry attempt through a port that is a direct connection to the Expansion Port Network receives a challenge response.

INADS

Indicates whether or not any entry attempt through a port that is a direct connection to the Initialization and Administration System (INADS) receives a challenge response. INADS is used to remotely initialize and administer Communication Manager

MGR1

Indicates whether or not any entry attempt through a port that is a direct connection to the system administration and maintenance access interface located on the processor circuit pack receives a challenge response.

NET

Indicates whether or not any entry attempt through a port that is a dialed-in or dialed-out connection to the Network Controller circuit pack receives a challenge response.

Translation-ID Number Mismatch Interval (days)

Valid Entry	Usage
1 to 90	The number of days the system allows access to system administration commands. When this interval expires, only init logins have the ability to execute system administration commands to modify translation data.

REMOTE MANAGED SERVICES

Port Board Security Notification

Enables or disables port board denial of service notification. Available only if **RMS Feature Enabled** is enabled.

Related Links

RMS Feature Enabled on page 664

Port Board Security Notification Interval

Valid Entry	Usage
60 to 3600 in increments of 10	The interval in seconds between port board Denial of Service notifications (traps). Default is 60. **Note:** There is no delay before the first trap is sent. The interval administered in this field applies only to the period between the sending of the traps. Available only if Remote Managed Services and Port Board Security Notification are enabled.

Related Links

Port Board Security Notification on page 663
REMOTE MANAGED SERVICES on page 663

RMS Feature Enabled

Enables or disables Remote Managed Services.

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Station Security Code Violation Notification Enabled

Use this field to enable or disable the security violation notification for Station Security Codes. Station Security Codes are used to validate logins to a particular extension.

The default value is n.

STATION SECURITY CODE VERIFICATION PARAMETERS

Minimum Station Security Code Length

Valid Entry	Usage
3 to 8	The minimum required length of the station security codes. Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be seven or eight.

Receive Unencrypted from IP Endpoints

Allows or blocks unencrypted data from IP endpoints.

Security Code for Terminal Self Administration Required

Specifies whether or not a Personal Station Access code is required to enter the Self-Administration mode.

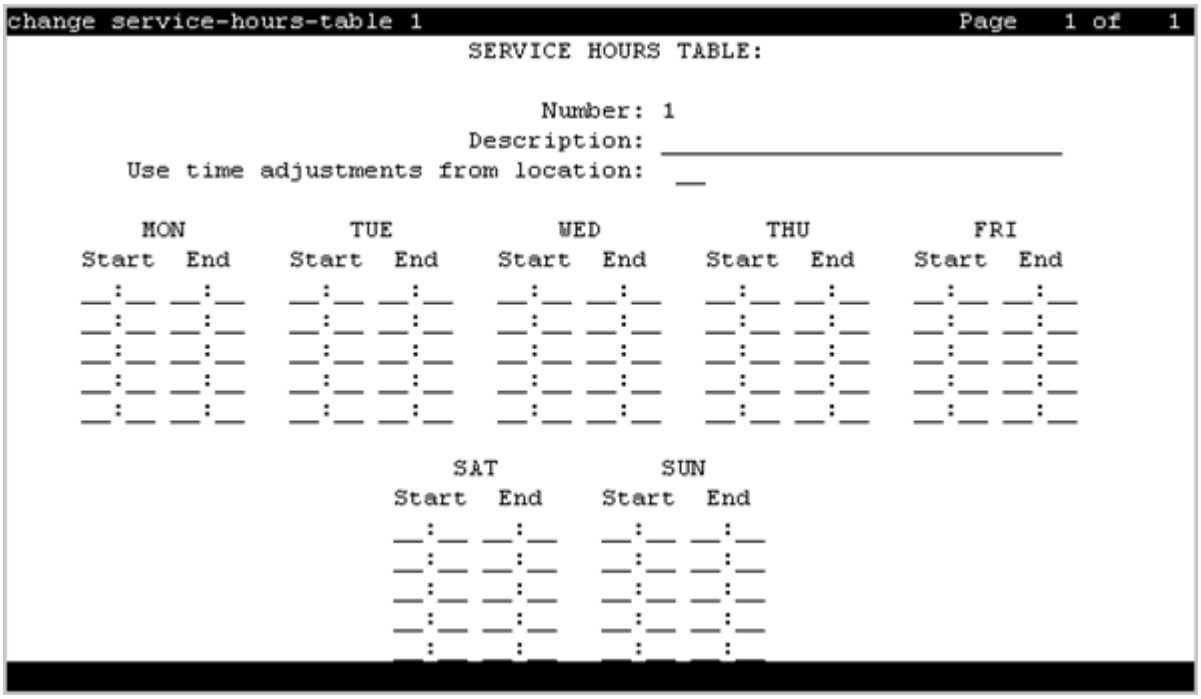
Related Links

Personal Station Access (PSA) on page 804

Chapter 113: Service Hours Table

Specifies office service hours using up to 99 different tables. Available only if basic vectoring is enabled.

Example command: `change service-hours-table n`, where *n* is the service hours table number.



Description

A name that provides a description of the table. Accepts up to 27 characters. The default is blank.

Example

Call-ahead Reservations

Number

The table number.

Start/End

The range of office hours for each day of the week.

A time is in the table from the first second of the start time (for example, 08:00:00) until the last second of the end time (for example, 17:00:59).

Valid Entry	Usage
0 to 23	Hour range. The hour range must be within the specified day, from 00:00 (midnight) until 23:59.
0 to 59	Minute range.

Use Time Adjustments from Location

The location number that specifies how time zone offset and daylight saving rule adjustments are performed.

Related Links

Loc Number on page 540

Chapter 114: Signaling group

Establishes signaling group parameters for ISDN-PRI, H.323, atm and SIP trunks. Because these trunk types vary in the types of parameters needed, the system displays these fields on this screen change depending on the **Group Type**.

Signaling group: page 1

Alert Incoming SIP Crisis Calls

This field is used to enable the Emergency call routing for H.323 visiting users feature.

The Emergency call routing for H.323 visiting users feature support crisis alert for H.323 visiting users. Emergency calls made by H.323 visiting users from an enterprise network are routed to the nearest possible emergency response agencies (PSAPs) at the location visited by the user.

This field is available only for SIP signaling groups.

Valid entry	Usage
y	The attendant endpoints are notified for the incoming emergency call.
n	The attendant endpoints are not notified for the incoming emergency call. The default value is n.

Alternate Route Timer

Available only if **Group Type** is sip.

Note:

You must the **LAR** field to a value other than none. If you set the **LAR** field to none, the call drops.

Valid Entry	Usage
2 to 30	The time in seconds Communication Manager waits for a Ringing message from the far end of the trunk before trying for an alternate route to establish a call. The default value is 6 seconds.

Related Links

Group Type on page 675

Associated Signaling

Available only if **Group Type** field is isdn-pri.

Valid Entry	Usage
y	Enables associated signaling.
n	Enables non-facility associated signaling.

Related Links

Group Type on page 675

Bypass If IP Threshold Exceeded

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
y	Automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below administered system-wide limits.
n	Signaling group trunks stay in service when IP transport performance falls below limits.

Related Links

IP options system parameters: page 1 on page 452
Group Type on page 675

Calls Share IP Signaling Connection

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
y	Enables inter-connection between servers running Avaya Communication Manager. Unavailable if a RAS-Location Request (LRQ) is enabled. **Note:** When the near and far-end servers are running Avaya Communication Manager, this value must match the value administered for enabling the Layer 3 test.
n	Disables inter-connection between servers running Avaya Communication Manager. Used if the local or remote server is *not* running Avaya Communication Manager.

Related Links

Group Type on page 675
LRQ Required on page 680
Enable Layer 3 Test on page 673

Circuit Type

Available only if **Group Type** is atm.

Valid Entry	Usage
T1	U.S., Canadian, and Japanese digital transmission format.
E1	European digital transmission format.

Related Links

Group Type on page 675

Connect

To control communications at layers 2 and 3 of the ISDN-PRI protocol, this field specifies what is on the far end of the link.

Available only if **Group Type** is atm.

Valid Entry	Usage
host	The link connects Communication Manager to a computer.
network	The link connects Communication Manager to a local telephone company central office or any other public network switch.
pbx	The link is connected to another switch in a private network.

Related Links

Group Type on page 675

Country Protocol

The country protocol used by the local telephone company central office where the link terminates. This value must match the country protocol used by the far-end server. For connections to a public network, the network service provider can tell you which country protocol they are using.

Available only if **Group Type** is atm.

Valid Entry	Usage
1 to 25	The country code.
etsi	The network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Used only if the **Signaling Mode** is isdn-pri.

Related Links

Signaling Mode on page 686
Country options table on page 783
Group Type on page 675

D Channel

Available only if the **Group Type** is atm.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks) 09 to 32	Six and seventh characters are the circuit number.

Related Links

Group Type on page 675

DCP/Analog Bearer Capability

Sets the information transfer capability in a bearer capability IE of a setup message to speech or 3.1kHz. Available only if **Group Type** is atm or h.323.

Valid Entry	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability. This is the default.
speech	Provides speech encoding in the information transfer capability.

Related Links

Group Type on page 675

Direct IP-IP Audio Connections

Allows or disallows direct audio connections between H.323 endpoints. Direct audio connections save bandwidth resources and improve sound quality of voice over IP (VoIP) transmissions. For SIP trunk groups, allows direct audio connections between SIP endpoints. Available only if **Group Type** is h.323 or sip.

Related Links

Group Type on page 675

DTMF over IP

The **DTMF over IP** field specifies the touchtone signals that are used for dual-tone multifrequency (DTMF) telephone signaling. The **DTMF over IP** field is available if the **Group Type** field is set to h.323 or sip.

Valid Entry	Usage
in-band	All G711 and G729 calls pass DTMF in-band. DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band. It also supports the SIP trunks.
in-band-g711	Only G711 calls pass DTMF in-band. The system displays the **in-band-g711** option if the **Group Type** field is set to h.323.
out-of-band	All IP calls pass DTMF out-of-band. For IP trunks, the digits are done with either Keypad IEs or H245 indications. This is the default for newly added H.323 signaling groups. The out-of-band option enables out of band signaling for the SIP Signaling group. When you select the out-of-band option, Communication Manager sends all outgoing DTMF messages as SIP INFO messages over a SIP signaling group. This option is interoperable with H.323 networks. You can connect an Avaya non-SIP endpoint or trunk to a voice mail system on the H.323 network that is integrated with Session Manager and Communication Manager through SIP. Examples of Avaya non-SIP endpoints or trunks are H.323, Analog, DCP, and Digital trunks.
rtp-payload	This is the method specified by RFC 2833. This is the default value for newly added SIP signaling groups. Support for SIP trunks requires the default entry of rtp-payload.

Related Links

Group Type on page 675

Enable Layer 3 Test

Enables or disables a Layer 3 test. Communication Manager runs the Layer 3 test to verify that all connections known at the near-end are recognized at the far-end. Available only if **Group Type** is h.323.

Note:

The Layer 3 test should be enabled for an H.323 signaling group type when the **Calls Share IP Signaling Connection** is enabled and the far-end is Communication Manager. Requires administration of a **Far-end Node Name**.

Related Links

Calls Share IP Signaling Connection on page 669
Far-end Node Name on page 675
Group Type on page 675

Enforce SIPS URI for SRTP

The system displays this when the **Group Type** field is sip. Use this field to enable or disable Communication Manager to enforce SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP.

Valid Entry	Usage
y	Enable Communication Manager to enforce SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP. Note: The system displays y as the default value for the **Enforce SIPS URI for SRTP** field.
n	Disable Communication Manager from enforcing SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP.

ETSI CCBS Support

Available only if **Group Type** is isdn-pri and **TSC Supplementary Service Protocol** is set to c for ETSI.

Valid Entry	Usage
none	Interface supports neither incoming nor outgoing ETSI CCBS. This is the default.
incoming	Interface supports only incoming ETSI CCBS.
outgoing	Interface supports only outgoing ETSI CCBS.
both directions	Interface supports incoming and outgoing ETSI CCBS.

Table continues…

Valid Entry	Usage
	Note: When upgrading from a version of Communication Manager that is earlier than 5.1, this is the default.

Related Links

TSC Supplementary Service Protocol on page 688
Group Type on page 675

Far-end Domain

The name of the IP domain that is assigned to the far-end of the signaling group. For example, to route Session Manager calls within an enterprise, the domain assigned to the proxy server is used. For external SIP calling, the domain name could be that of the SIP service provider. Available only if **Group Type** is sip.

Valid Entry	Usage
Character string	A maximum of 40 characters used to define the name of the IP domain for which the far-end proxy is responsible, if different than the near-end domain.
blank	**Far-end domain** is unspecified and the far-end IP address is used. Use if the domains are the same.

Related Links

Group Type on page 675

Far-end Listen Port

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
1 to 65535	Must match the number administered for **Near-end Listen Port**. Typically, this is the default of 5061 for SIP over TLS.
blank	Far-end listen port is unspecified.

Related Links

Near-end Listen Port on page 681
Group Type on page 675

Far-end Network Region

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
1 to 2000	The number of the network region that is assigned to the far-end of the signaling group. The region is used to obtain the codec set used for negotiation of trunk bearer capability.
blank	Selects the region of the near-end node and the far-end network region is unspecified.

Related Links

Group Type on page 675

Far-end Node Name

Use this field to specify the node name for the far-end IP interface that is used for trunks assigned to the Signaling group. You must administer the node name on the Node Names screen.

The system displays the field only if the **Group Type** field is set to h.323 or sip.

Related Links

Node Name on page 429
Group Type on page 675

Group Number

The signaling group number.

Group Type

The type of protocol used with the signaling group.

Valid Entry	Usage
h.323	h.323 protocols or SBS signaling trunks
isdn-pri	Integrated Service Digital Network Primary Rate Interface
sip	For SIP

H.235 Annex H Required

Enables or disables the requirement that the Communication Manager server uses the H.235 Annex H (now called H.235.5) protocol for authentication during registration. Available only if **Group Type** is h.323. Requires that RAS-Location Request (LRQ) is enabled.

Related Links

LRQ Required on page 680

Group Type on page 675

H.245 DTMF Signal Tone Duration (msec)

Available only if **DTMF over IP** is set to out-of-band .

Valid Entry	Usage
80 to 350 blank	Specifies the duration of DTMF tones sent in H.245-signal messages. The default is blank.

Related Links

DTMF over IP on page 672
Group Type on page 675

H.323 Station Outgoing Direct Media

Available if **Group Type** is h.323 and **Direct IP-IP Audio Conections** is enabled. Also available if **Group Type** is SIP and **Direct IP-IP Audio Conections** is enabled.

Valid Entry	Usage
y	A call from an H.323 station over a trunk that uses this signaling group starts as a direct media call. The IP address and port of the H.323 station are sent as the media and media control channel addresses in the SETUP/INVITE message. **Note:** On an outgoing Direct Media call from an IP (H.323) telephone over this trunk, if an attempt is made to transfer the call, conference another party, or put the call on hold while the call is still in ringing state, the operation fails.
n	The IP address of the MEDPRO board is sent in the SETUP/INVITE message. This is the default.

Related Links

Direct IP-IP Audio Connections on page 672
Group Type on page 675

Idle Code

An eight-digit string that is compatible with the protocol used by the far-end switch or server. Sets the signal sent out over idle DS0 channels.

Available only if **Group Type** is atm.

Related Links

Group Type on page 675

IMS Enabled

Enables or disables accepting SIP requests that match the domain in the **Far-End Domain**. This field enables the signaling group to use the IMS signaling. Available only if **Group Type** is sip. If you are configuring Communication Manager as a feature server, set the **IMS Enabled** field to y. Setting this field to y only applies when Communication Manager is being configured as a feature server. If you are configuring Communication Manager as an evolution server, leave the **IMS Enabled** field set to n.

Related Links

Far-end Domain on page 674
Group Type on page 675

Incoming Dialog Loopbacks

The system displays this on the Signaling Group screen when the **Group Type** field is sip.

Valid entries	Usage
allow	Communication Manager software connects the call and allows the SIP trunks to remain in the looparound connection. If the trunk group controlled by this SIP signaling group is used for IGAR calls, set the value allow.
eliminate	Communication Manager software connects the call and eliminates the SIP trunks from the looparound connection. The default value is eliminate.

Initial IP-IP Direct Media

The system displays the **Initial IP-IP Direct Media** field on the Signaling Group screen only when you set the **Group Type** field to sip and the **Direct IP-IP Audio Connections** field to y. The **Initial IP-IP Direct Media** field enables or disables the SIP direct media feature between SIP endpoints. The **Initial IP-IP Direct Media** field also enables or disables the direct media feature from SIP endpoints to Avaya H.323 stations. The default value for this field is n.

Valid Entry	Usage
n	Disables the SIP direct media feature. The default value is n.
y	Enables the SIP direct media feature and SIP-H.323 direct media feature.

Related Links

Direct IP-IP Audio Connections on page 672
Group Type on page 675

Interface

Controls how your server negotiates glare with the far-end switch. Available only for a far-end PBX connection.

Valid entry	Usage
U.S. private network applications	
network	The server overrides the other end when glare occurs. For servers connected to a host computer.
user	The server releases the contested circuit and looks for another when glare occurs. For servers connected to a public network.
Private networks, including QSIG networks, applications outside the U.S.	
peer-master	The switch overrides the other end when glare occurs.
peer-slave	The switch releases the contested circuit and looks for another when glare occurs.

Related Links

Connect on page 670

Interface Companding

The companding algorithm expected by the system at the far end.

Valid Entry	Usage
a-law	Algorithm expected at the far-end for E1 service.
mu-law	Algorithm expected at the far-end for T1 service.

Interworking Message

Determines what message Communication Manager sends when an incoming ISDN trunk call is routed over a non-ISDN trunk group.

Valid Entry	Usage
PROGress	Requests the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk. Normally-selected value.
ALERTing	Causes the public network in many countries to play ringback tone to the caller. This value is used only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

IP Audio Hairpinning

Enables or disables hairpinning for H.323 or SIP trunk groups. H.323 endpoints are connected through the IP circuit pack without going through the time division multiplexing (TDM) bus. Available only if **Group Type** is h.323 or sip.

Related Links

Group Type on page 675

IP Video

Enables or disables IP video capability for this signaling group. Available only if the signaling group type h.323 and sip.

Link Loss Delay Timer (sec)

Specifies how long to hold the call state information in the event of an IP network failure or disruption. Communication Manager preserves calls and starts this timer at the onset of network disruption (signaling socket failure). If the signaling channel recovers before the timer expires, all call state information is preserved and the signaling channel is recovered. If the signaling channel does not recover before the timer expires, the system:

- Raises an alarm against the signaling channel
- Maintains all connections with the signaling channel
- Discards all call state information about the signaling channel

Valid Entry	Usage
1 to 180	The number of seconds to delay the reaction of the call controller to a link bounce. Default is 90.

Location for Routing Incoming Calls

The system displays this only when the **Group Type** field on the Signaling Group screen is h.323 and **Multiple Locations** field on the Optional Feature screen is y.

Valid Entry	Usage
1 to 2000	(Depending on your server configuration, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.) Determines the location-based digit analysis which is applied to the destination digits dialed by the incoming trunk call.

Table continues…

Valid Entry	Usage
blank	The location of CLAN's cabinet or the location of the Processor Ethernet (PE) is taken for routing the call, depending on whether CLAN or PE is being used to carry that H.323 signaling channel. By default, the value is blank.

LRQ Required

Enables or disables RAS-Location Request (LRQ) messages. Available only if **Group Type** is h.323.

Valid Entry	Usage
y	Allows IP trunk availability to be determined on a per call basis. An LRQ message is sent to the far-end gatekeeper prior to each call over the IP trunk. The far-end gatekeeper responds with a RAS-Location Confirm (LCF) message, and the call proceeds. Required if **H.235 Annex H Required** is enabled. **Calls Share IP Signaling Connection** must be disabled.
n	For far-end servers running Communication Manager, unless **H.235 Annex H Required** is enabled.

Related Links

H.235 Annex H Required on page 675
Calls Share IP Signaling Connection on page 669
Group Type on page 675

Max number of CA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
0 to 619	The maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.

Related Links

Group Type on page 675

Max number of NCA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
0 to 256	The maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Related Links

Group Type on page 675

Media Encryption

Enables or disables encryption for trunk calls assigned to this signaling group. If encryption for the signaling group is disabled, then trunk calls using this signaling group do not get encrypted regardless of IP Codec Set administration. Available only if Media Encryption is enabled in Communication Manager and the **Group Type** is h.323.

Related Links

Group Type on page 675

Name

A name that identifies the signaling group. Accepts up to 15 alphanumeric characters.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Group Type on page 675

Near-end Listen Port

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
5000 to 9999	An unused port number for the near-end listen port.
1719	For LRQ-enabled ports
1720	For h.323-enabled ports

Related Links

H.323 Station Outgoing Direct Media on page 676
LRQ Required on page 680
Group Type on page 675

Near-end Node Name

The node name for the Control LAN (C-LAN) IP interface in the Avaya S8XXX Server. The node name must be previously administered.

Available only if **Group Type** is h.323 or sip.

Related Links

Name on page 451
Node Name on page 429
Group Type on page 675

Network Call Transfer

Enables or disables Network Call Transfer so that D-channels support Explicit Network Call Transfer (ENCT). Available only if **Group Type** is atm.

Related Links

Group Type on page 675

Passphrase

The passphrase used to generate a shared “secret” for symmetric encryption of the media session key. The same passphrase must be assigned to the corresponding signaling groups at both ends of an IP trunk. The passphrase:

- Consists of 8 to 30 alphanumeric characters
- Is case sensitive
- Must contain at least one alphabetic and at least one numeric
- Valid characters also include letters, numbers, and these symbols: !&*?;’^(),.:-

The passphrase is used for both Media Encryption and authentication. This field cannot be left blank.

Available only if Media Encryption is enabled or the **H.235 Annex H Required** is enabled.

Related Links

H.235 Annex H Required on page 675

Peer Detection Enabled

The values for this field are `n(o)` and `y(es)`. The default value is `y(es)`. This field is only available if **Group Type** is `sip`. When set to `y(es)` then Communication Manager will automatically set the value in the **Peer Server** field when the sip signaling group comes into service.

Peer Server

The values for this field are `Others` and `SM`. The default value is `Others`. This field is only available if **Group Type** is `sip`. If this sip signaling group terminates to an Avaya Session Manager, then Communication Manager will automatically set value in this field to `SM` if the **Peer Detection Enabled** field is set to `y(es)`.

Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers

If you set the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field on the SIP Signaling Group screen to y and if Communication Manager uses the public-unknown-numbering table, Communication Manager inserts a leading plus (+) digit in calling, alerting, diverting, or connected numbers. The default value is n.

If the **Peer Server** field on the SIP Signaling Group screen is SM, the system sets the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field to display-only, with the default value of y. If the **Peer Server** field on the SIP Signaling Group screen is Others, the system sets the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field to n and as editable.

Primary D Channel

Available only if **Group Type** is isdn-pri.

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth character are the slot number.
01 to 04 (Analog TIE trunks) 09 to 32	Six and seventh characters are the circuit number.

Related Links

Group Type on page 675

Priority Video

Enables or disables the priority video function that specifies that incoming video calls have an increased likelihood of receiving bandwidth, and are also allocated a larger maximum bandwidth per call.

Available only if:

- **Group Type** is h.323 or sip
- Multimedia SIP trunking is enabled for the system

Related Links

Multimedia IP SIP Trunking on page 803
Group Type on page 675

Protocol Version

Available only if **Group Type** is atm.

Valid Entry	Usage
a b c d	In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service. Selects the protocol that matches the network service provider's protocol.

Related Links

Group Type on page 675

Q-SIP

The system displays this only when the **Group Type** field is h.323 or sip.

Valid Entry	Usage
y	Enables the QSIG over SIP (Q-SIP) feature for the signaling group.
n	The QSIG over SIP feature is disabled. By default, the value is n.

QSIG Signaling Group

The system displays this only when the **Group Type** field is sip and the **Q-SIP** field is set to y.

Valid Entry	Usage
1 to 999	Assigns a number for the QSIG signaling group.
blank	No QSIG signaling group is assigned. By default, the value is blank.

Remove '+' from Incoming Called/Calling/Alerting/Diverting/ Connected Numbers

If you set the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field on the SIP Signaling Group screen to y, Communication Manager removes the leading plus (+) digit, if present, from both calling and called numbers. The default value is y.

This field interacts with the **Peer Server** field in the following manner:

- If the **Peer Server** field on the SIP Signaling Group screen is SM, the system sets the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field to n. The **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field is editable.
- If the **Peer Server** field on the SIP Signaling Group screen is Others, the system sets the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field to y. The **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field is editable.

Remote Office

Enables or disables administering this signaling group for a remote office. Available only if **Group Type** is h.323.

Related Links

Group Type on page 675

RFC 3389 Comfort Noise

The system displays this on the Signaling Group screen when the **Group Type** field is sip.

Valid entries	Usage
y	This enables SIP signaling for comfort noise. If **RFC 3389 Comfort Noise** field is set to y, this field overrides the **Silence Suppression** field on the IP Codec Set screen.
n	This disables SIP signaling for comfort noise. Default is n.

RRQ Required

Enables or disables the requirement that a vendor registration be sent. Available only if **Group Type** is h.323.

Related Links

Group Type on page 675

SBS

Enables or disables the Separation of Bearer and Signaling (SBS) trunk groups. If this field is enabled, both the **Trunk Group for NCA TSC** and the **Trunk Group for Channel Selection** must be set to the signaling group number administered for the SBS trunk group. Available only if **Group Type** is set to h.323.

Related Links

Trunk Group for NCA TSC on page 688
Trunk Group for Channel Selection on page 687
Group Type on page 675

Session Establishment Timer (min)

Available only if **Group Type** is sip.

Valid Entry	Usage
3 to 120	The time in minutes Communication Manager waits before tearing down a ring no answer call. The default is 3 minutes.

Related Links

Group Type on page 675

Signaling Mode

Displays the isdn-pri signaling mode. Available only if **Group Type** is atm.

Related Links

Group Type on page 675

SIP Signaling Group

The system displays this only when the **Group Type** field is h.323 and the **Q-SIP** field is set to y.

Valid Entry	Usage
1 to 999	Assigns a number for the SIP signaling group.
blank	No SIP signaling group is assigned. By default, the value is blank.

T303 Timer (sec)

Available only if **Group Type** is h.323.

Valid Entry	Usage
2 to 10	The number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Default is 10.

Related Links

Group Type on page 675

Transport Method

Available only if **Group Type** is sip.

Valid Entry	Usage
tcp	Transport is accomplished using Transmission Control Protocol (TCP).
tls	Transport is accomplished using Transport Layer Security (TLS). This is the default.

Related Links

Group Type on page 675

Trunk Group for Channel Selection

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
1 to 2000	Trunk group number used for channel selection.

Related Links

Group Type on page 675

Trunk Group for NCA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
1 to 2000 blank	Trunk group number used for non-call-associated Temporary Signaling Connections.

Related Links

Group Type on page 675

TSC Supplementary Service Protocol

The supplementary service protocol used for temporary signaling connections.

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
a	AT&T, Telcordia Technologies, Nortel.
b	ISO QSIG. Also, for SBS signaling groups.
c	ETSI. Available only if **Group Type** is isdn-pri.
d	ECMA QSIG
e	Allows DCS with rerouting. **DCS with Rerouting** must be enabled, and **Used for DCS** must be enabled for the trunk group.
f	Feature Plus
g	ANSI. Available only if **ISDN-PRI** , or **ISDN-BRI**, or **Used for DCS** are enabled for the system.

Related Links

DCS with Rerouting on page 797
Used for DCS on page 911
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801
Group Type on page 675

Virtual Channel Identifier

Available only if **Group Type** is atm.

Valid Entry	Usage
32 to 1023 blank	The number used as a Virtual Channel Identifier.

Related Links

Group Type on page 675

Virtual Path Identifier

Available only if **Group Type** is atm.

Valid Entry	Usage
32 to 1023 blank	The number used as a Virtual Path Identifier.

Related Links

Group Type on page 675

X-Mobility/Wireless Type

The type of X-Mobile endpoints allowed. Available only if **Group Type** is isdn-pri.

Valid Entry	Usage
DECT	The remote end of the trunk group controlled by the signaling group is a DECT mobility controller. This allows X-Mobility to work over ISDN-PRI trunks between the server/switch and adjunct.
none	Not administered.

Related Links

Group Type on page 675

Signaling group: page 2

This screen must be filled in for ATM signaling groups. It provides two functions:

- Defines fractional T1 and fractional E1 facilities, specifying how many and which channels to use.
- Specifies the port numbers to use. Port numbers must be unique for all signaling boards on the same ATM board.

Available only if **Group Type** is atm.

LIMIT SIGNALING GROUP USAGE

The LIMIT SIGNALING GROUP USAGE screen is available only if the **Group Type** field is set to h.323 or sip, and the **Near-end Node Name** field is set to procr. Use this screen to control where H.323 and SIP trunks are used.

Related Links

Group Type on page 675
Near-end Node Name on page 682

Enable on Survivable Processors (ESS and LSP)

Valid Entry	Usage
all	To enable the signaling group and the associated trunks on both - Survivable core server (Enterprise Survivable Server (ESS)) and Survivable remote server (Local Survivable Processor (LSP)).
ess-all	To enable the signaling group and the associated trunks on Survivable core server (Enterprise Survivable Server) but not on Survivable remote server (Local Survivable Processor).
none	To block the signaling group and the associated trunks on survivable processors and free system resources.
selected	To specify which survivable processors can use the signaling group and the associated trunks.

Enable on the Main Processor(s)

Use the **Enable on the Main Processor(s)** field to enable or disable the signaling group and the associated trunks on the main server.

Valid Entry	Usage
y	To enable the signaling group and the associated trunks on the main server.
n	To disable the signaling group and the associated trunks on the main server.

Selected Survivable Processor Node Names

The **Selected Survivable Processor Node Names** field is available only if you set the **Enable on Survivable Processors (ESS and LSP)** field to selected. Use this field to specify the node names of the survivable processors that can use the signaling group and the associated trunks.

Related Links

Enable on Survivable Processors (ESS and LSP) on page 690

SIGNALING GROUP

Chan Port

Valid Entry	Usage
009 to 256	The port number for non-signaling channels.
blank	The signaling channel (port 16 for an E1 and port 24 for a T1) must be a port between 9 and 32. A port number used here cannot be used on any other ATM signaling group on the same board. The channels used must match exactly the channels used on the other end of the signaling group. For example, if your T1 is set up to use channels 1 through 5, 7, and 24 (the signaling channel), the far end must also use channels 1 through 5, 7, and 24.

Enable Failover Event Package Subscription

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **Multiple Level Precedence & Preemption** field on the System parameters customer options screen to `y`.

For Communication Manager to send and receive failover notify messages, you must configure **CCA-ID** on the Multiple Level Precedence & Preemption screen.

Note:

MLPP is a customer option. Therefore, you can change the parameters only through a license change.

Valid Entry	Usage
y	Displays the option request parameters on the screen.
n	Does not display the option request parameters on the screen. The default value is `n`.

Failover/failback Signaling-group Pair

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is

available only for the USA Department of Defense (DoD) and approved Federal government customers.

Type the Signaling group that connects to the primary Multi-Function Soft Switch (MFSS) or secondary MFSS.

A lower signaling group number is always the number assigned to the primary MFSS from the pair of signaling groups.

Valid Entry	Usage
SIP signaling-group number	The number that connects to the far-end MFSS.
Blank	The field is blank by default.

OPTIONS Request Parameters

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The system displays the OPTIONS Request Parameters fields on the screen only when you set the **Enable Failover Package Subscription** field to `y`.

Communication Manager creates and sends periodic OPTIONS messages to the MFSS. OPTIONS Request Parameters on the Signaling group screen monitor the interval between OPTIONS messages.

Interval between OPTIONS messages (seconds)

The interval time for Communication Manager to send OPTIONS requests to MFSS.

You can enter a number from 30 to 900 in increments of 10. The default value is `60`.

Number of OPTIONS message failure before failover

The number of OPTIONS messages that Communication Managersends to the far-end MFSS without getting a response. A failover is triggered when the far-end MFSS does not respond to the OPTIONS messages.

You can enter a number from 1 to 10. The default value is `2`.

Number of OPTIONS message successes before failback

The number of successful OPTIONS messages that Communication Manager sends to the far-end MFSS. A failback is triggered when the far-end MFSS starts responding to the OPTIONS messages.

You can enter a number from 1 to 10. The default value is `2`.

Wait time before failback (seconds)

The time taken for Communication Manager to failback to the primary MFSS after the far-end MFSS starts responding to the OPTIONS messages.

You can enter 1 or any number from 10 to 250 in increments of 10.

Signaling group: NCA-TSC Assignment page

Appl.

Specifies the application for this administered NCA-TSC.

Valid Entry	Usage
audix	The ISDN-PRI D-channel DCS Audix feature
dcs	The DCS Over ISDN-PRI D-channel feature.
gateway	The administered NCA-TSC is used as one end in the gateway channel.
masi	The NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Message conversion from an administered AUDIX NCA-TSC to a QSIG CISC. A Machine ID between 1 and 20 is required.

As-needed Inactivity Time-out (min)

Valid Entry	Usage
10 to 90 blank	Applies only to as-needed NCA-TSCs.

Dest. Digits

An ISDN interface extension number. Accepts up to 15 characters including * and #.

Enabled

Enables or disables the administered NCA-TSC.

Established

The strategy for establishing this administered NCA-TSC.

Valid Entry	Usage
permanent	The administered NCA-TSC can be established by either the near end or the far end.
as-needed	The administered NCA-TSC is established the first time the administered NCA-TSC is needed. The NCA-TSC can be set up either by the near end or far end switch.

Local Ext

The local extension of the ISDN interface.

Mach ID

Valid Entry	Usage
1 to 63 for DCS, 1 to 99 for AUDIX, 1 to 15 for MASI, and 1 to 63 for QSIG-MWI	A unique machine ID. You cannot specify an ID that you already administered for the Processor Channel.

Related Links

Mach ID on page 636

Service/Feature

Valid Entry	Usage
accunet i800 inwats lds mega800 megacom multiquest operator sdn	The assigned service or feature. In addition to pre-defined services or features, any user-defined Facility Type of 0 (feature) or 1 (service) is allowed.

Valid Entry	Usage
sub-operator wats-max-band	

Chapter 115: SIT Treatment for Call Classification

The treatment of Special Information Tones (SITs) used for Outbound Call Management type calls with USA tone characteristics. The port network TN744 Call Classifier circuit pack ports or Branch Gateway internal tone detector resources in classified mode are used to detect SITs. The classifiers are capable of detecting the following SITs:

- SIT Ineffective Other
- SIT Intercept
- SIT No Circuit
- SIT Reorder
- SIT Vacant Code
- SIT Unknown
- AMD (Answering Machine Detected) Treatment

Available only if **ASAI Link Core Capabilities** and **ASAI Link Plus Capabilities** are enabled for the system.

Example command: `change sit-treatment`

```
change sit-treatment                                          Page   1 of   1
                     SIT TREATMENT FOR CALL CLASSIFICATION


                       SIT Ineffective Other: dropped
                              SIT Intercept: answered
                             SIT No Circuit: dropped
                                SIT Reorder: dropped
                            SIT Vacant Code: dropped
                                SIT Unknown: dropped

                               AMD Treatment: dropped
                  Pause Duration (seconds): 1.0
                   Talk Duration (seconds): 1.5
```


Comments on this document? infodev@avaya.com

Related Links

ASAI Link Core Capabilities on page 795
ASAI Link Plus Capabilities on page 795

AMD Treatment

Specifies the treatment for Answering Machine Detected. An ASAI adjunct can request AMD for a call. If an Answering Machine is detected, one of two treatments is specified.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

Pause Duration

Fractions of a second pause duration, as opposed to **Talk Duration** that is for full seconds.

Valid Entry	Usage
0.1 to 2.0 seconds in increments of 0.1 seconds	The amount of time Communication Manager looks for a pause before it classifies the call as a live person. The Pause Duration timer should be set longer than the typical silence between words in an answering machine greeting, but shorter than the typical space between words in a live greeting. Defaults to 0.5 seconds.

SIT Ineffective Other

The treatment for “Ineffective Other” messages. An example of an Ineffective Other message is “You are not required to dial a 1 when calling this number”.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Intercept

The treatment for intercept messages that direct callers to another number or extension.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent. This is the default.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent.

Example

An example of an intercept message is “XXX-XXXX has been changed to YYY-YYYY, make a note of it”.

SIT No Circuit

The treatment for calls with circuit problems, such as busy.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Reorder

The treatment for reorder calls.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Unknown

The treatment for calls when a situation or condition that is unknown to the network is encountered.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Vacant Code

The treatment for calls when a vacant code is encountered.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

Talk Duration

Full second talk duration, as opposed to **Pause Duration** that is for fractions of a second.

Valid Entry	Usage
0.1 to 5.0 seconds in increments of 0.1 seconds	The amount of time Communication Manager looks for voice energy. If it finds that much continuous speech, Communication Manager classifies the call as an answering machine. The Talk Duration timer should be set to a time longer than it takes to say a typical live greeting. Defaults to 2.0 seconds.

Chapter 116: Site Data

Provides descriptive information about the buildings, floors and telephone set colors. Values must be supplied here before site data can be entered for the station.

Chapter 117: Station

Administers individual telephone sets or virtual telephones. Some of the fields are used for specific telephone types, while others are used for all telephone types.

1-Step Clearing

Enables or disables call termination at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List 1, List 2, List 3

Assigns up to three abbreviated dialing lists to each telephone.

Valid Entry	Usage
enhanced	Telephone user can access the enhanced system abbreviated dialing list.
group	Telephone user can access the specified group abbreviated dialing list. Requires administration of a group number.
personal	Telephone user can access and program their personal abbreviated dialing list. Requires administration of a personal list number.
system	Telephone user can access the system abbreviated dialing list.

Access Code

A five-digit access code used to place a wireless terminal into service. The access code is a temporary, shorter version of the complete User Authentication Key (UAK) required by the system when the terminal is first put into service. It is used to automatically generate a unique UAK for that wireless terminal over-the-air.

Available only if a wireless terminal model number is selected as the station type.

Active Station Ringing

Defines how calls ring to the telephone when it is off-hook without affecting how calls ring at this telephone when the telephone is on-hook.

Valid Entry	Usage
continuous	All calls to this telephone ring continuously.
single	Calls to this telephone receive one ring cycle and then ring silently.
if-busy-single	Calls to this telephone ring continuously when the telephone is off-hook and idle. Calls to this telephone receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent	All calls to this station ring silently.

Adjunct Supervision

Available only if the station type is 500, 2500, k2500, 8110, ops, ds1fd, ds1sa, VRU, VRUFD, or VRUSA.

Valid Entry	Usage
y	An analog disconnect signal is sent automatically to the port after a call terminates. Analog devices, such as answering machines and speakerphones, use this signal to turn the devices off after a call terminates.
n	Required hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Always Use

Use this field to enable the following emergency call handling settings:

- A softphone can register irrespective of the emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the value administered in the **Emergency Location Extension** field is used as the calling party number. The user-entered emergency call handling settings of the softphone are ignored.
- If an IP telephone dials 911, the value administered in the **Emergency Location Extension** field is used as the calling party number.
- If an agent dials 911, the physical station extension is used as the calling party number, overriding the value administered in the **LoginID for ISDN Display** field.

Does not apply to SCCAN wireless telephones, or to extensions administered as type H.323.

Assigned Member — Ext

The extension of the user who has an associated **Data Extension** button and shares the module.

Assigned Member — Name

The name associated with the extension of the user who has an associated **Data Extension** button and shares the module.

Att. Call Waiting Indication

With attendant call waiting, attendant-originated or attendant-extended calls to a busy single-line telephone can wait and sends distinctive call-waiting tone to the single-line user.

Valid Entry	Usage
y	Activates Call Waiting for the telephone without Caller ID information. This feature must be enabled when the **Type** is set to H.323. This is the default.
n	Call Waiting is disabled for the station. Disable this feature if: • **Data Restriction** is enabled • **Switchhook Flash** field is disabled • **Data Privacy** is enabled for the telephone's class of service (COS)
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. Available only if the station **Type** is CallrID.

Audible Message Waiting

Enables or disables an audible message waiting tone indicating the user has a waiting message consisting of a stutter dial tone when the user goes off-hook.

This field does *not* control the Message Waiting lamp.

Available only if **Audible Message Waiting** is enabled for the system.

AUDIX Name

The voice messaging system associated with the station. Must contain a user-defined adjunct name that was previously administered.

Auto-A/D

Enables or disables automatic abbreviated/delayed ringing for a call appearance. Available only if **Per Button Ring Control** is enabled.

Auto Answer

In an Expert Agent Environment (EAS) environment, the auto answer setting for an Agent LoginID overrides the endpoint settings when the agent logs in.In EAS environments, the auto answer setting for the Agent LoginID can override a station's setting when an agent logs in.

Valid entry	Usage
all	All ACD and non-ACD calls to an idle station cut through immediately. The agent cannot use automatic hands-free answer for intercom calls. With non-ACD calls, the station rings while the call is cut through. To prevent the station from ringing, activate the **ringer-off** feature button, provided the Allow Ringer-off with Auto-Answer feature is enabled for the system.
acd	Only ACD split, ACD skill, and direct agent calls cut through. Non-ACD calls to the station ring audibly. For analog stations: • Only the ACD split or skill calls and direct agent calls cut through. • Non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the agent receives call-waiting tone.
none	All calls to the station receive an audible ringing.
icom	The user can answer an intercom call from the same intercom group without pressing the intercom button.

Automatic Moves

With Automatic Moves, you can unplug a DCP telephone from one location and moved to a new location without additional Communication Manager administration. Communication Manager automatically associates the extension to the new port.

Caution:

When a DCP telephone is unplugged and moved to another physical location, the **Emergency Location Extension** must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid Entry	Usage
always	The DCP telephone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	The DCP telephone can be unplugged and plugged into a new location once. After a move, the field is set to done the next time that routine maintenance runs on the DCP telephone. Use when moving a large number of DCP telephones so each extension is removed from the move list. Also use to prevent automatic maintenance replacement.
no	Requires administration to move the DCP telephone.
done	Communication Manager sets the field to done after the telephone is moved and routine maintenance runs on the DCP telephone.
error	Communication Manager sets the field to error, after routine maintenance runs on the DCP telephone, when a non-serialized telephone is set as a movable telephone.

Auto Select Any Idle Appearance

Enables or disables automatic selection of any idle appearance for transferred or conferenced calls. Communication Manager first attempts to find an idle appearance that has the same extension number as the call being transferred or conferenced has. If that attempt fails, Communication Manager selects the first idle appearance.

Automatic Selection of DID Numbers

Enables or disables the Automatic Selection of DID Numbers for Guest Rooms feature. This feature assigns a two- to five-digit extension from a predetermined list of numbers to a hotel room telephone number that is disassociated with the room number.

BCC

Indicates voice or voice-grade data when the value is set to 0. The Bearer Capability Classes (BCC) value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking). Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Bearer

Used when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. The system the **Bearer** field for 8503, 8510, and 8520 stations in Communication Manager 2.1 and 2.2 only. **Secure Terminal Equip** is for Bearer functionality in Communication Manager 3.0 and later.

Valid Entry	Usage
speech	Forces the Bearer Cap IE to “speech” before a call is delivered to the 85*xx* BRI station.
3.1khz	Leaves the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Bridged Appearance Origination Restriction

Restricts or allows call origination on the bridged appearance.

Valid Entry	Usage
y	Call origination on the bridged appearance is restricted.
n	Call origination ion the bridged appearance is allowed. This is normal behavior, and is the default.

Bridged Call Alerting

Controls how the user is alerted to incoming calls on a bridged appearance.

Valid Entry	Usage
y	The bridged appearance rings when a call arrives at the primary telephone.
n	The bridged appearance flashes but does not ring when a call arrives at the primary telephone. This is the default.

Table continues…

Valid Entry	Usage
	If disabled and **Per Button Ring Control** is also disabled, audible ringing is suppressed for incoming calls on bridged appearances of another telephone's primary extension.

Bridged Idle Line Preference

Use this field to specify that the line that the system selects when you go off hook is always an idle call appearance for incoming bridged calls.

Valid entry	Usage
y	The user connects to an idle call appearance instead of the ringing call.
n	The user connects to the ringing bridged appearance.

Building

A valid building location.

Busy Auto Callback without Flash

Enables or disables automatic callback for a calling analog station without flashing the hook. Available for analog telephones only if **Without Flash** is enabled for the system.

BUTTON ASSIGNMENTS

The feature assigned to each button on the station. Feature buttons are assigned by entering the abbreviated feature. For a list of feature buttons, see Telephone Feature Buttons Table in *Administering Avaya Aura® Communication Manager* , 03-300509.

Note:

To use Terminal Translation Initialization (TTI), a call appearance (call-appr) must be assigned to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

Identifies the cable that connects the telephone jack to the system.

Call Appearance Display Format

Specifies the display format for the station. Bridged call appearances are not affected by this field. This field is available only on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones.

Note:

This field sets the administered display value only for an individual station.

Valid Entry	Usage
loc-param-default	The system uses the administered system-wide default value. This is the default.
inter-location	The system displays the complete extension on downloadable call appearance buttons.
intra-location	The system displays a shortened or abbreviated version of the extension on downloadable call appearance buttons.

Caller ID Message Waiting Indication

Allows or prevents aliasing of various non-Avaya telephones and adjuncts. Available only if **Type** is CallrID. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Note:

The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON-lamp Communication Manager Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID telephone with **Caller ID Message Waiting Indication** is disabled and **Message Waiting Indicator** set to neon.

Calls Allowed

Identifies the Extension to Cellular call filter type for an XMOBILE station. This field allows an XMOBILE station to function as a bridge and still be restricted. Available only with an EC500-type XMOBILE station and with a **Mapping Mode** of termination or both.

Valid Entry	Usage
internal	External calls are blocked. Internal calls terminate to the XMOBILE station. Attendant-originated and attendant-delivered calls are not delivered
external	Internal calls are blocked. External calls terminate to the XMOBILE station.
all	All calls terminate to the XMOBILE station.
none	Prevents calls from terminating to the XMOBILE station. Can be used to prevent business-related calls from accruing telephone charges on cellular telephones that are lost, being transferred to a new user, or being disabled for other business reasons.

Note:

Interswitch calls on DCS trunks are treated as internal calls. When this field is set to:

- internal or all, DCS calls are delivered to the cell telephone
- external or none, DCS calls are not delivered

Incoming calls from other Extension to Cellular users are internal if office caller ID is enabled for the XMOBILE station associated with the cell telephone. When this field is set to:

- internal or all, calls from other Extension to Cellular users are delivered
- external or none, calls from other Extension to Cellular users are not delivered

Call Waiting Indication

User, attendant-originated, and outside calls to a busy single-line telephone can wait and sends a distinctive call-waiting tone to the single-line user. Not available if **Data Restriction** is enabled, **Switchhook Flash** is disabled, or if Data Privacy is active by way of the telephone COS assignment.

Valid Entry	Usage
y	Activates Call Waiting without Caller ID information for the telephone. This is the default.
n	Call Waiting is disabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the station type is CallrID.

CDR Privacy

Enables or disables Call Privacy for each station. With CDR Privacy, digits in the called number field of an outgoing call record can be blanked on a per-station basis. The number of blocked digits is administered system-wide as CDR parameters.

Cell Phone Number

The unformatted cell telephone's published external number consisting of 1 to 15 digits. This field can contain a three-digit area code plus the seven-digit main number. If the same Cell Phone Number is administered for multiple XMOBILE stations, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same. This number consist of a full 10-digit Cell Phone Number regardless of whether the cell telephone is local or not.

Configuration Set

Valid Entry	Usage
1 to 10	The configuration set number that contains the call treatment options required for the XMOBILE station. This field must be administered if: • The **XMOBILE Type** is EC500. • The **Mobility Trunk Group** is a trunk group number and the administered trunk group is non-DECT or non-PHS. • The **Mobility Trunk Group** is aar or ars.
blank	If the **Mobility Trunk Group** is a trunk group number and the administered trunk group is DECT or PHS, this field can be left blank.

Conf/Trans On Primary Appearance

Enables or disables the forced use of a primary appearance when the held call to be conferenced or transferred is a bridge. This is regardless of the administered value for **Auto Select Any Idle Appearance**.

COR

Class of Restriction (COR) number with the required restriction.

Cord Length

The length of the cord attached to the receiver. This is a free-form entry, and can be in any measurement units.

COS

The Class of Service (COS) number used to select allowed features.

Country Protocol

The protocol that corresponds to the supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Coverage After Forwarding

Governs whether an unanswered forwarded call is provided coverage treatment.

Valid Entry	Usage
y	Coverage treatment is provided after forwarding regardless of the administered system-wide coverage parameters.
n	No coverage treatment is provided after forwarding regardless of the administered system-wide coverage parameters.
s(ystem)	Administered system-wide coverage parameters determine treatment.

Coverage Module

Indicates whether or not a coverage module is connected to the station.

Coverage Msg Retrieval

Allows or denies users in the telephone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this telephone. Applies only if the telephone is enabled for LWC Reception.

Coverage Path 1 or Coverage Path 2

The coverage-path number or time-of-day table number assigned to the station.

Note:

If Modified Misoperation is active, a Coverage Path must be assigned to all stations on Communication Manager.

CRV Length

Valid Entry	Usage
1 or 2	The length of the Call Reference Value (CRV) for each interface. Only for ASAI stations.

Custom Selection of VIP DID Numbers

The selection or removal of a DID number assigned to a room when a guest checks in. Available only if **Automatic Selection of DID Numbers** is enabled.

Related Links

Automatic Selection of DID Numbers on page 360

Customizable Labels

Enables or disables the Increase Text for Feature Buttons feature for this station. This feature expands the text labels associated with Abbreviated Dial buttons from the current five uppercase alphanumeric characters to a maximum of 13 upper and lower case alphanumeric characters. Ensures that there will always be sufficient customized button resources to support VIP users. Available only when the station type is one of the following:

- 2410 (Release 2 or later)

- 2420 (Release 4 or later)
- 4610 (IP Telephone Release 2.2 or later)
- 4620 (IP Telephone Release 2.2 or later)
- 4621 (IP Telephone Release 2.2 or later)
- 4622 (IP Telephone Release 2.2 or later)
- 4625 (IP Telephone Release 3.1 or later)

Data Extension

Use this field to assign an extension to the data module. You can enter up to five digits. This value must conform to the dial plan.

Data Module

Indicates whether or not this telephone has an associated data module.

Data Option

Valid Entry	Usage
analog	A second line on the telephone is administered on the I-2 channel.
data module	A second line on the telephone is *not* administered on the I-2 channel.
none	The data option is not administered.

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Default Dialing Abbreviated Dialing Dial Code

The list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number.

Available only if the **Special Dialing Option** is set to default.

Dial Prefix

The unformatted sequence of up to four digits or characters that are prepended to the cell telephone's published cell telephone number before dialing. Accepts the * and # characters. If the same Cell Phone Number is administered on multiple XMOBILE stations, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

Display Caller ID

Enables or disables transmission of calling party information to the Caller ID telephone or adjunct. For CallrID type telephones or analog telephones with Caller ID adjuncts only. Available only if the station type is CallrID.

Display Cartridge

Enables or disables displaying the cartridge associated with the station. For 7404 D telephones only.

Display Client Redirection

Enables or disables the display of redirection information for a call originating from a station with Client Room Class of Service and terminating to this station. When disabled, only the client name and extension or room display. Available only if Hospitality is enabled for the system.

Note:

This field must be enabled for stations administered for any type of voice messaging that needs display information.

Display Language

Valid Entry	Usage
english french italian spanish user-defined	The language that displays on stations. Time of day is displayed in 24-hour format (00:00 - 23:59) for all languages except English, which is displayed in 12-hour format (12:00 a.m. to 11:59 p.m.).
unicode	Displays English messages in a 24-hour format. If no Unicode file is installed, displays messages in English by default. **Note:** Unicode display is only available for Unicode-supported telephones. Currently, 4610SW, 4620SW, 4621SW, 4622SW, 16xx, 96xx, 96x1, and 9600-series telephones (Avaya one-X Deskphone Edition SIP R2 or later) support Unicode display. Unicode is also an option for DP1020 (aka 2420J) and SP1020 (Toshiba SIP Phone) telephones when enabled for the system.

Distinctive Audible Alert

Use this field to enable the distinctive audible alerts to allow telephones to receive three different types of ringing patterns that identify the type of incoming calls. Distinctive ringing might not work properly for off-premises telephones.

DSN Number

You can enter a 10-digit extension, or leave the field blank.

Emergency Location Ext

The Emergency Location Extension for this station. This extension identifies the street address or nearby location when an emergency call is made. Defaults to the telephone's extension. Accepts up to thirteen digits.

Note:

On the ARS Digit Analysis Table in Communication Manager, 911 must be administered to be call type emer or alrt for the E911 Emergency feature to work properly.

Related Links

Remote Softphone Emergency Calls on page 733

EMU Login Allowed

Enables or disables using the station as a visited station by an Enterprise Mobility User (EMU).

Endpt ID

Available only if Endpt Init is enabled.

Valid Entry	Usage
00 to 62	A unique two-digit number for this endpoint. Each Endpt ID field must have a unique value for each endpoint on the same port.

Endpt Init

Indicates the terminal's endpoint initialization capability. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made through the SPID, administered into the system, and entered into the ISDN-BRI terminal.

Available only if **MIM Support** is enabled.

Valid Entry	Usage
y	The terminal supports Telcordia Technologies ISDN-1 terminal initialization procedures.
n	For all other country protocols.

Enhanced Callr-Info display for 1-line display phones

Use this field to enable or disable the **Callr-Info Persists** feature. Use this feature with telephones that have a one-line display and a **callr-info** button that displays the collected digits. For more information on **Callr-info Persists**, see *Avaya Aura® Call Center Elite Feature Reference*.

Valid entry	Usage
n	This entry is the default. If an agent presses the **callr-info** button, the telephone displays the callr-info digits. The callr-info digits are cleared after: • the timeout period, defined by the **Callr-info Display Timer** field, expires or • the agent disconnects the call
y	On receiving an incoming call with callr-info digits associated, the indicator of the **callr-info** button flashes. • If the agent presses the **callr-info** button, the caller information is displayed on the telephone and the status lamp of the button becomes steady. If the agent presses the **callr-info** button after the telephone displays the caller information, the telephone displays the caller information again and the indicator of the **callr-info** button winks. • If the agent does not press the button, the status lamp continues to flash during the call.

Expansion Module

Indicates whether or not this telephone has an expansion module. Enables the administration of the buttons for the expansion module.

Extension

The extension for this station.

For a virtual extension, a valid physical extension or a blank can be entered. Using Blank, an incoming call to the virtual extension can be redirected to the virtual extension “busy” or “all” coverage path.

Feature Module

Indicates whether or not the station is connected to a feature module.

Fixed TEI

Indicates whether or not the endpoint has a fixed Terminal Endpoint Identifier (TEI). The TEI identifies a unique access point within a service. TEIs must be administered for fixed TEI terminals. Must be enabled for ASAI.

Available only for ISDN-BRI data modules, NI-BRI telephones, WCBRI data modules, and ASAI links.

Floor

A valid floor location.

Forwarded Destination

A destination extension for both internal and external calls for each of the three types of enhanced call forwarding (Unconditional, Busy, and No Reply). Accepts up to 18 digits. The first digit can be an asterisk *.

Requires administration to indicate whether the specific destination is active (enabled) or inactive (disabled).

H.320 Conversion

Use this field to enable the conversion of H.320-compliant calls to voice-only calls for the attendant console.

Note:

The system can handle only a limited number of conversion calls. Therefore, the number of attendant consoles with H.320 conversion must be limited.

Headset

Indicates whether or not the telephone has a headset.

Home

Indicates the roaming status of the wireless user. Available only when a wireless terminal model number is selected as the station type.

Valid Entry	Usage
y	The user's home. This is the default.
n	The roaming system.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Available only if **Special Dialing Option** is hot-line.

Hot Line Service is used when very fast service is required and when a telephone is used only for accessing a certain facility.

HOT LINE DESTINATION — Abbreviated Dialing List Number

The abbreviated dialing list where the hotline destination number is stored.

HOT LINE DESTINATION — Dial Code

The dial code in the specified abbreviated dialing list where the hotline destination number is stored.

Hunt-to Station

The extension the system must hunt to for this telephone when the telephone is busy. You can create a station hunting chain by assigning a hunt-to station to a series of telephones.

Idle/Active Ringing (Callmaster)

Defines how a call rings to the telephone when it is on-hook. Applies to CALLMASTER telephones.

Valid Entry	Usage
continuous	All calls to this telephone ring continuously.
if-busy-single	Calls to this telephone ring continuously when the telephone is off-hook and idle, and calls to this telephone receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent-if-busy	Calls ring silently when this station is busy.
single	Calls to this telephone receive one ring cycle and then ring silently.

Idle Appearance Preference

Indicates which call appearance is selected when the user lifts the handset and there is an incoming call.

Valid Entry	Usage
y	The user connects to an idle call appearance instead of the ringing call.
n	The Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

Indicates whether or not rotary digits from the set are ignored. If enabled, the short switch-hook flash (50 to 150) from a 2500-type set is ignored.

IPEI

Available when a wireless terminal model number is selected as the station type.

Valid Entry	Usage
0 to 9 a to f A to F blank	The International Portable Equipment Identifier of the wireless terminal. Accepts a unique nine-character hexadecimal ID number.

IP Audio Hairpinning

Indicates whether or not IP endpoints are allowed to connect through the IP circuit pack in the server without going through the time division multiplexing (TDM) bus.

Available only if the station type is one of the several h.323 or sip types.

IP Phone Group ID

Available only for H.323 station types.

Valid Entry	Usage
0 to 999 blank	The Group ID number for this station.

IP Softphone

Indicates whether or not this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

Available only for DCP station types and IP Telephones.

IP Video

Indicates whether or not this extension has IP video capability. Available only for station type h.323.

IP Video Softphone

Indicates whether or not this extension is a video softphone. Available only if **IP Softphone** is enabled.

ITC (Information Transfer Capability)

The type of transmission facilities used for ISDN calls originating from this endpoint. Unavailable for voice-only or BRI stations.

Valid Entry	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission. In other words, a sequence of eight digital zeros are converted to a sequence of seven zeros and a digital one.
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission. In other words, digital information is sent exactly as is.

Jack

Alpha-numeric identification of the jack used for this station.

Location

The system displays this field only when you set the **Multiple Locations** field on the system parameters customer options screen to y, and set the **Type** field to H.323 or SIP station types.

Valid entry	Usage
1 to 2000	(Depending on your server configuration, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.) Assigns the location number to a particular station. Allows IP telephones and softphones connected through a VPN to be associated with the branch an employee is assigned to. This field is one way to associate a location with a station. For the other ways and for a list of features that use location, see the Location sections in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.
blank	Indicates that the existing location algorithm applies. By default, the value is blank.

Lock Messages

Controls access to voice messages by other users.

Valid Entry	Usage
y	Restricts other users from reading or canceling the voice messages, or retrieving messages using Voice Message Retrieval.
n	Allows other users to read, cancel, or retrieve messages.

Loss Group

Valid Entry	Usage
1 to 17	Determines which administered two-party row in the loss plan applies to each station. Is not displayed for stations that do not use loss, such as x-mobile stations.

LWC Activation

Activates or deactivates the Leave Word Calling (LWC) feature. With LWC, internal telephone users on this extension can leave short pre-programmed messages for other internal users.

You must use LWC if:

- The system has hospitality and the guest-room telephones require LWC messages indicating that wakeup calls failed
- The LWC messages are stored in a voice-messaging system

LWC Log External Calls

Determines whether or not unanswered external call logs are available to end users. When external calls are not answered, Communication Manager keeps a record of up to 15 calls provided information on the caller identification is available. Each record consists of the latest call attempt date and time.

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related Links

AUDIX Name on page 389

Mapping Mode

Controls the mode of operation in which the cell telephone operates when mapped to this XMOBILE extension. An XMOBILE station can be bridged to a deskset. These restrictions or modes exist because the COR of a bridge is ignored; instead the principal's COR is used. With Mapping Mode, an XMOBILE station can function as a bridge and still be restricted.

When a cell telephone is mapped to more than one XMOBILE station, then only one of the mapped XMOBILE stations can have origination or both as its Mapping Mode. Therefore, only one of the XMOBILE stations mapped to the cell telephone number is permitted to originate calls.

Valid Entry	Usage
both	The cell telephone can be used to originate and terminate calls from its associated XMOBILE extension. This is the default when the XMOBILE type is PHS or DECT.
none	The XMOBILE station is disabled administratively and cannot originate and terminate calls from its associated internal extension.
origination	The cell telephone can be used only to originate calls from its associated internal XMOBILE extension by dialing into the office server running Communication Manager.
termination	The cell telephone can be used only to terminate calls from its associated internal XMOBILE extension. This is the default when the XMOBILE type is EC500.

Map-to Station

The extension of a physical telephone used for calls to a virtual extension. Cannot be used with an xmobile, xdid or any other virtual extension.

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Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

Valid Entry	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Lamp Ext

The extension of the station tracked with the message waiting lamp.

Message Server Name

Specifies which Message Server is associated with the station. Must contain a user-defined adjunct name that was previously administered. Names must be administered in alphabetical order.

Message Waiting Indicator

Specifies the type of message waiting indicator. This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID telephones. Must be set to a value other than none when the station type is set to H.323.

Available only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, H.323 and VRU telephones.

Valid Entries	Usage
led	The message waiting indicator is a light-emitting diode (LED).
neon	The message waiting indicator is a neon indicator.

Table continues…

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Valid Entries	Usage
	Note: The neon message waiting indicator is supported only on a small subset of boards, including older US-only boards, such as the TN746 and the TN793. Check the documentation for your board to see if neon is supported. Available only if the **Analog Ringing Cadence** is set to 1 (US) for the location parameters.
none	No message waiting indicator is selected. This is the default.

Message Waiting Type

Valid Entry	Usage
DISPL	The MW text is added to the end of the display line during an incoming or outgoing call.
ICON	The MWI message is used to update to the Wireless Terminal.
NONE	No MWI message is sent to the Wireless Terminal

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other than endpoint initialization. Available only if **MIM Support** is enabled.

MIM Support (Management Information Message Support)

Enables or disables MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities. Available only for ISDN-BRI data modules and ASAI. Must be disabled for ASAI.

Mobility Trunk Group

Associates the XMOBILE station to a trunk.

Valid Entries	Usage
2000	A valid trunk group number for mobility routing. This trunk group is used for routing.

Table continues…

Valid Entries	Usage
aar	The Automatic Alternate Routing (AAR) capabilities of Communication Manager are used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call is not extended out of the server. It provides ringback to the calling company and might eventually go to coverage.
ars	The Automatic Route Selection (ARS) capabilities of Communication Manager are used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call is not extended out of the server. It provides ringback to the calling company and might eventually go to coverage.
blank	Not administered.

Model

The model of the NI-BRI telephone.

Valid Entry	Usage
L-3 Communication STE	The NI-BRI telephone is a model L-3 Communication STE.
Tone Commander	The NI-BRI telephone is a model 6210 and 6220 Tone Commander.
Other	The NI-BRI telephone is another model (for example, a Nortel 5317T).

Mounting

Indicates whether the station mounting is d(esk) or w(all).

Multimedia Early Answer

Enables or disables multimedia early answer on a station-by-station basis.

You must enable the station for the Multimedia Early Answer feature if the station receives coverage calls for multimedia complexes, but is not multimedia-capable. This ensures that calls are converted and the talk path is established before ringing at this station.

Mute Button Enabled

Enables or disables the mute button on the station.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid Entries	Usage
fp-mwi	The station is a served user of an fp-mwi message center.
qsig-mwi	The station is a served user of a qsig-mwi message center.
blank	The served user's MWI is not audited or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

The name of the person associated with this telephone or data module. The system uses this value to create the system directory.

Note:

This field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW telephones.

For more information on Unicode language display, see "Administering Unicode Display" in *Administering Avaya Aura® Communication Manager* .

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Note:

In the display for emergency notification when completing the **Name** field, fill the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the **Name** field appear on the first display line, followed by the extension. The second line contains the last three characters of the **Name** field, followed by the word `EMERGENCY`. Characters 18 through 24 of the **Name** field are not displayed at all.

Offline Call Logging

Use the **Offline Call Logging** field to back up and restore call logs for an offline user.

Valid entry	Usage
y	Communication Manager tracks the calls that the user missed when the telephone was in the logged-out state, and updates the call logs on the telephone when the user logs in. The default option is y.
n	Communication Manager does not track the calls that the user missed when the telephone was in the logged-out state.

Off Premises Station

Available only for analog telephones.

Valid Entries	Usage
y	This telephone is located in a different building with the system. Requires administration of R Balance Network.
n	The telephone is located in the same building with the system.

Passive Signalling Station

The entries y and n indicate the call processing status of the analog trunk port user. This field is available only when you select an analog trunk port as the station type.

Valid Entry	Usage
y	The analog station port does not process calls when off hook.
n	The analog station port processes and receives calls as an analog station. This is the default value.

PCOL/TEG Call Alerting

Enables or disables alerting at the station for Personal CO Line/Terminating Extension Group calls.

Available only for 510 telephones.

Per Button Ring Control

Enables or disables per button ring control by the station user.

Valid Entries	Usage
y	Users can select ring behavior individually for each call-appr, brdg-appr, or abrdg-appr on the station and to enable Automatic Abbreviated and Delayed ring transition for each call-appr on the station. Prevents the system from automatically moving the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Calls on **call-appr** buttons always ring the station and calls on **brdg-appr** or **abrdg-appr** buttons always ring or not ring based on the **Bridged Call Alerting** value. The system can move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Defines the personalized ringing pattern for the station. Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. For virtual stations, this field dictates the ringing pattern on its mapped-to physical telephone.

L = 530 Hz, M = 750 Hz, and H = 1060 Hz

Valid Entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Determines Calling Party Number (CPN) information sent on outgoing calls from this station.

Valid Entries	Usage
y	All outgoing calls from the station deliver the CPN information as “Presentation Allowed.”
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station delivers the Calling Party Number information as “Presentation Restricted.”
blank	The sending of CPN information for calls is controlled by administration on the outgoing trunk group the calls are carried on.

Port

The Auxiliary and Analog ports assigned to the station are as follows.

Valid Entry	Usage
01 to 64	The first and second numbers are the cabinet numbers.
A to E	The third character is the carrier.
01 to 20	The fourth and fifth characters are the slot numbers. G650 has 14 slots.
01 to 32	The sixth and seventh characters are the port numbers.
x or X	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension has a non-IP set. Or, the extension had a non-IP set, and it dissociated. Use x for Administered WithOut Hardware (AWOH) and Computer Telephony (CTI) stations, as well as for SBS Extensions.
IP	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have an IP set. This is automatically entered for certain IP station set types, but you can enter for a DCP set with softphone permissions. This changes to the s00000 type when the set registers.
xxxVmpp	Specifies the Branch Gateway. • xxx is the Branch Gateway number, which is in the range 001 to 250. • m is the module number, which is in the range 1 to 9. • pp is the port number, which is in the range 01 to 32.
Analog Trunk port	Analog trunk port is available with: • MM711 and MM714 media modules • TN747 and TN797 circuit packs

Precedence Call Waiting

Activates or deactivates Precedence Call Waiting for this station.

Public (PSTN) Number

You can enter an 11-digit extension, or leave the field blank.

R Balance Network

You must enter a value in this field if you set the **Off-Premise Station** field to y.

Valid Entry	Usage
y	Selects the Resistor Only Balance network. The Resistor Only Balance network is used when Communication Manager is connected to terminal equipment, such as an SLC carrier or an impedance compensator, that provides an input impedance of 600 ohm and Communication Manager is less than 3,000 feet from the equipment.
n	Selects the Resistor and Capacitor (RC) balance network. RC balance network, when connected to off-premise stations, provide better signal transmission than Resistor only balance network.

Recall Rotary Digit

Enables or disables the Recall Rotary Digit dialing that enables this user to perform conference and transfer operations.

Available only if the station type is 500 or 2500.

Redirect Notification

Enables or disables redirection notification that gives a half ring at this telephone when calls to this extension are redirected through Call Forwarding or Call Coverage. Must be enabled if LWC messages are stored on a voice-messaging system.

Remote Office Phone

Indicates whether or not this station is used as an endpoint in a remote office configuration.

Remote Softphone Emergency Calls

Tells Communication Manager how to handle emergency calls from the IP telephone.

Caution:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. You cannot use an Avaya IP endpoint to dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. Avoid using an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. If you have questions about emergency calls from IP telephones, go to the Avaya Support website at http://support.avaya.com.

Available only if the station is an IP Softphone or a remote office station.

Valid Entry	Usage
as-on-local	If the emergency location extension that corresponds to this station's IP address is not administered (left blank), the value as-on-local sends the station emergency location extension to the Public Safety Answering Point (PSAP). If the administrator populates the IP address mapping with emergency numbers, the value as-on-local functions as follows: • If the station emergency location extension is the same as the IP address mapping emergency location extension, the value as-on-local sends the station's own extension to the Public Safety Answering Point (PSAP). • If the station emergency location extension is different from the IP address mapping emergency location extension, the value as-on-local sends the IP address mapping extension to the Public Safety Answering Point (PSAP).
block	Prevents the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the server than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.

Table continues…

Valid Entry	Usage
cesid	Allows Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the server that an emergency call reaches the PSAP that covers the softphone's physical location. If the server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Allows the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. This entry is used for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between block and cesid on the softphone. A DCP or IP telephone in the office automatically selects the extension.

Require Mutual Authentication if TLS

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The TLS server, Communication Manager, and the TLS client, 96x1 H.323 phone, must both exchange Identity certificates and mutually authenticate the other side certificate.

If an H.323 phone has a certificate and the **Require Mutual Authentication if TLS** field is set to:

- `y`: Communication Manager validates the certificate. If the certificate is valid, Communication Manager allows TLS setup with H.323 phones. If the certificate is invalid, Communication Manager rejects the TLS request.
- `n`: Communication Manager does not check for the phone certificate. Communication Manager allows TLS setup with H.323 phones to proceed without an endpoint certificate.

If an H.323 phone does not have a certificate and the **Require Mutual Authentication if TLS** field is set to:

- `y`: Communication Manager does not accept the security profile, H323TLS, in the Gatekeeper Confirmation (GCF). If the endpoint attempts to set up TLS, Communication Manager rejects the TLS request.
- `n`: Communication Manager allows TLS setup with H.323 phones to proceed without an endpoint certificate.

Restrict Last Appearance

Valid Entries	Usage
y	Restricts the last idle call appearance used for incoming priority calls and outgoing call originations only.
n	Last idle call appearance is used for incoming priority calls and outgoing call originations.

Rg

Valid Entry	Usage
a(bbreviated-ring) d(elayed-ring) n(o-ring) r(ing)	The type of automatic abbreviated/delayed ringing for each call appearance when per button ring control is enabled. Default is r.

Room

Valid Entry	Usage
Telephone location	Identifies the telephone location. Accepts up to 10 characters.
Guest room number	Identifies the guest room number if this station is one of several to be assigned a guest room and the **Display Room Information in Call Display** is enabled for the system. Accepts up to five digits.

SAC/CF Override

With **SAC/CF Override**, the user of the calling station can override the redirection set by the called station.

Valid entry	Usage
ask	The system prompts the user of the calling station whether the call must follow the redirection path or override the redirection path. The user can type y or n.
no	The user of the calling station cannot override the redirection path of the call. The call follows the redirection path.

Table continues…

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Valid entry	Usage
yes	The user of the calling station can override the redirection path of the call, provided the called station has at least one idle call appearance.

Secure Terminal Equip

Used when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. Available only for 8503, 8510, and 8520 stations in Communication Manager 3.0 and later.

Valid Entries	Usage
y	Leave the Bearer Cap IE unchanged. 3.1khz is used to let secure calls from Secure Terminal Equipment (STE) telephones work properly.
n	Force the Bearer Cap IE to “speech” before a call is delivered to the 85*xx* BRI station.

Security Code

The security code required by users for specific system features and functions are as follows:

- Extended User Administration of Redirected Calls
- Personal Station Access
- Redirection of Calls Coverage Off-Net
- Leave Word Calling
- Extended Call Forwarding
- Station Lock
- Voice Message Retrieval
- Terminal Self-Administration
- Enterprise Mobility User
- Extension to Cellular
- Call Forwarding
- Posted Messages
- Security Violation Notification
- Demand Printing

The required security code length is administered system wide.

Select Last Used Appearance

Valid Entry	Usage
y	Indicates a station's line selection is not to be moved from the currently selected line button to a different, non-alerting line button. The line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	The line selection on an on-hook station with no alerting calls can be moved to a different line button that might be serving a different extension.

Service Link Mode

Use this field to specify the duration of a service link connection. The service link is the combined hardware and software multimedia connection between an H.320 Desktop Video Conferencing (DVC) system and Communication Manager.

The service link is established when a user receives or makes a call during a multimedia, IP softphone, or IP telephone session.

Valid entry	Usage
as-needed	For multimedia, IP softphone, and IP telephone users. The service link remains connected for 10 seconds after the user disconnects a call so that the user can immediately make or receive another call. After 10 seconds, the link is disconnected, and a new link must be established to make or receive a call.
permanent	For call center agents who are constantly making or receiving calls during the multimedia, IP softphone, or IP telephone session. The service link remains connected for the entire duration of the session.

Set Color

Indicates the set color. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

You can change the list of allowed set colors by using the Valid Set Color fields on the site-data screen.

Short/Prefixed Registration Allowed

If the **IP Stations** or the **IP Softphone** field is set to y, the system displays the **Short/Prefixed Registration Allowed** field for this H.323 IP set types: 1603, 1608, 1616, 4601, 4601P, 4602, 4602P, 4606, 4610, 4612, 4620, 4622, 4624, 4630, 9610, 9620, 9630, and 9650.

Valid Entry	Usage
y	Call processing allows the short registration to proceed per station basis. This entry overrides the System Parameters IP Options screen field value for that station.
n	Call processing rejects the short registration per station basis. This entry overrides the System Parameters IP Options screen field value for that station.
default	Call processing uses the **Short/Prefixed Registration Allowed** field option on the System Parameters IP Options screen for short registration. The default value is default.

Related Links

Short/Prefixed Registration Allowed on page 453

Signaling Type

The **Signaling Type** field is used to specify how the A and B bits are used on an Off Premise Station (OPS) line.

Valid Entry	Usage
Fx	The signaling bits are manipulated as if the port is terminating on a Foreign Exchange (FX) unit in a D4 channel bank. This value is default.
Inverted	The signaling bits are manipulated as if the port is terminating on a Special Access Unit (SAU) in a D4 channel bank.

SIP Trunk

The system displays this field on SIP station types. This field shows the value of the Trunk Selection of the corresponding off-pbx-telephone station mapping entry if there is one. The system displays an error message and also blocks the administration of the field if the user tries to enter a value that is not the trunk group number of an existing SIP trunk, ARS or AAR.

Speaker

Indicates whether or not the station is equipped with a speaker.

Speakerphone

Controls the behavior of speakerphones.

Valid Entry	Usage
1-way	Indicates that the speakerphone listen-only.
2-way	Indicates that the speakerphone is both talk and listen.
grp-listen	With Group Listen, a telephone user can talk and listen to another party with the handset or headset while the telephone's two-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party through the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party. Available only with 6400-series and 2420/2410 telephones.
none	Not administered for a speakerphone.

Special Dialing Option

Identifies the type of dialing for calls when this data module originates calls.

Valid Entry	Usage
hot-line	The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.
default	The system dials the default dialing AD number.
blank	Normal keyboard dialing.

SPID — (Service Profile Identifier)

The Service Profile IDentifier (SPID) for this station. The SPID is a variable length parameter required for ISDN-BRI stations.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port is blocked.

Note:

If you have not administered a port for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Available only if the terminal supports ISDN terminal initialization procedures.

Survivable COR

Sets a level of restriction for stations to be used with the survivable dial plan to limit certain users to only to certain types of calls. You can list the restriction levels in order from the most restrictive to least restrictive. Each level has the calling ability of the ones above it. This field is used by PIM module of the Integrated Management to communicate with the Communication Manager administration tables and obtain the class of service information. PIM module builds a managed database to send for Standard Local Survivability (SLS) on the Branch Gateways.

Available for all analog and IP station types.

Valid Entries	Usage
emergency	This station can only be used to place emergency calls.
internal	This station can only make intra-switch calls. This is the default.
local	This station can only make calls that are defined as locl, op, svc, or hnpa in the Survivable Gateway Call Controller's routing tables.
toll	This station can place any national toll calls that are defined as fnpa or natl on the Survivable Gateway Call Controller's routing tables.
unrestricted	This station can place a call to any number defined in the Survivable Gateway Call Controller's routing tables. Those strings marked as deny are also denied to these users.

Survivable GK Node Name

Any valid previously-administered IP node name. Identifies the existence of other H.323 gatekeepers located within gateway products that offer survivable call features. For example, the MultiTech MVPxxx-AV H.323 gateway family and the SLS function within the Branch Gateways. When a valid IP node name is entered into this field, Communication Manager adds the IP address of this gateway to the bottom of the Alternate Gatekeeper List for this IP network region. As H.323 IP stations register with Communication Manager, this list is sent down in the registration confirm

message. With this, the IP station can use the IP address of this Survivable Gatekeeper as the call controller of last resort.

If blank, there are no external gatekeeper nodes within a customer's network. This is the default value.

Available only if the station type is an H.323 station for the 46*xx* or 96*xx* models.

Survivable Trunk Dest

Designates certain telephones as not being allowed to receive incoming trunk calls when the Branch Gateway is in survivable mode. This field is used by the PIM module of the Integrated Management to successfully interrogate the Communication Manager administration tables and obtain the class of service information. PIM module builds a managed database to send for SLS on the Branch Gateways.

Available for all analog and IP station types.

Valid Entry	Usage
y	Allows this station to be an incoming trunk destination while the Branch Gateway is running in survivability mode. This is the default.
n	Prevents this station from receiving incoming trunk calls when in survivable mode.

Switchhook Flash

Valid Entry	Usage
y	Allows users to use the switchhook flash function to activate Conference/Transfer/ Hold and Call Waiting. Required for H.323 station types.
n	Disables the flash function so that when the switchhook is pressed while active on a call, the call drops. Requires that Call Waiting Indication is disabled.

TEI

Available only if the station has a fixed Terminal Endpoint Identifier (TEI).

Valid Entry	Usage
0 to 63	A one- or two-digit number that identifies the TEI of the station.

Tests

Valid Entry	Usage
y	Enables port maintenance tests.
n	Disables port maintenance tests. Required if the equipment (dictaphone) connected to the port does not support these tests.

Time of Day Lock Table

Valid Entry	Usage
1 to 5	Assigns the station to a Time of Day (TOD) Lock/Unlock table. The assigned table must be administered and active.
blank	Indicates no TOD Lock/Unlock feature is active. This is the default.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Type

Use this field to specify the type of telephone. You must administer the station type for each station that you add to the system.

The following table lists the telephones, virtual telephones, and personal computers that you can administer on Communication Manager. Telephones that are not in the table require an alias to a supported set type.

Note:

If TTI is enabled for the system, analog telephones administered with hardware to a virtual extension cannot be changed.

For more information about current documentation, product notices, knowledge articles, go to the Avaya Support website at http://support.avaya.com/.

Telephone type	Model	Administer as
Single-line analog	500	500

Table continues…

Telephone type	Model	Administer as
	2500, 2500 with Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multi-appearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H
Multi-appearance digital	2402	2402
	2410	2410
	2420	2420

Table continues…

Telephone type	Model	Administer as
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
Multi-appearance digital	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	9404	9404
	9408	9408
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1

Table continues…

Telephone type	Model	Administer as
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+ **Note:** When you add a new 4601 IP telephone, you must use the 4601+ station type. This station type enables the Automatic Callback feature.	4601+
	4602+ **Note:** When you add a new 4602 IP telephone, you must use the 4602+ station type. This station type enables the Automatic Callback feature.	4602+
	4606	4606
	4610	4610
	4612	4612
	4620SW IP (G3.5 hardware)	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
	9608	9608
	9610	9610
	9611	9611
	9620	9620
	9621	9621
	9630	9630
	9640	9640
	9641	9641
	9650	9650
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware	4620SIP

Table continues…

Telephone type	Model	Administer as
	• 4620SIP with SIP firmware • 4620SIP CC (Call Center) • SIP Softphone/Avaya one-X Desktop • Toshiba SP-1020A **Note:** You must administer any telephone that has SIP firmware and that is used for SIP networking as a 4620SIP telephone, 96xxSIP telephone, or 16CC SIP telephone. **Note:** Communication Manager does not support 1616SIP CC and 4620SIP CC telephones from Release 6.2 and later.	
	Avaya one-X (tm) Deskphone 9620, 9630, 9630G 9640, 9640G with SIP firmware	96xx or 96xxSIP telephone
	9608 with SIP firmware	9608SIP
	9611 with SIP firmware	9611SIP
	9621 with SIP firmware	9621SIP
	9641 with SIP firmware	9641SIP
	9608 with SIP firmware (for call center)	9608SIPCC
	9611 with SIP firmware (for call center)	9611SIPCC
	9621 with SIP firmware (for call center)	9621SIPCC
	9641 with SIP firmware (for call center)	9641SIPCC
H.323 SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	Any NI-BRI (N1 and N2) telephone	NI-BRI
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T

Table continues…

Telephone type	Model	Administer as
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		• Use XDID when Communication Manager assigns a DID number to the station • Use XDIDVIP when the administrator later assigns a DID number to the station • Use virtual to map the station and other extensions to one physical telephone
Key telephone system interface	—	K2500
ASAI	asai link computer telephony adjunct link	asai adjlk
AWOH	any digital set	same as Multi-appearance Digital
	CTI station	CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs

Type of 3PCC Enabled

Indicates if the telephone can be controlled by an Avaya 3rd Party Call Control (3PCC) adjunct or a Computer Telephony Integration (CTI) adjunct. Default is none.

Voice mail Number

Valid Entries	Usage
digits (1 to 9) * # ~p (pause)	Provides a voice mail retrieval AUX dial button on 4xx, 46xx, and 96xx telephones. When a number is entered in this field, the telephone's fixed voice mail retrieval button acts as an autodial button, dialing the number entered in this field to access voice mail. Accepts up to 24 digits.

Valid Entries	Usage
~w/~ (wait) ~m (mark) ~s (suppress)	**Note:** If this field is left blank, the telephone's fixed voice mail retrieval button acts as a “transfer to voice mail” button that only works for Intuity Audix or QSIG-integrated voice mail systems. For Intuity Audix and QSIG-integrated voice mail systems, leave the field blank. For non-QSIG integrated voice mail systems, set the field to an appropriate number.

XID

Identifies Layer 2 XID testing capability. Available only for an ISDN-BRI data module or an ASAI link.

Telephone type	Model	Administer as
Single-line analog	500	500
	2500, 2500 with Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa

Table continues…

Telephone type	Model	Administer as
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multi-appearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H
Multi-appearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
Multi-appearance digital	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+

Table continues…

Telephone type	Model	Administer as
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	9408	9408
	9404	9404
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+ **Note:** When adding a new 4601 IP telephone, you must use the 4601+ station type. This station type enables the Automatic Callback feature.	4601+
	4602+ **Note:** When adding a new 4602 IP telephone, you must use the 4602+ station type. This station type enables the Automatic Callback feature.	4602+
	4606	4606
	4610	4610
	4612	4612

Table continues…

Telephone type	Model	Administer as
	4620SW IP (G3.5 hardware)	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
	9611	9611
	9621	9621
	9630	9630
	9641	9641
	9650	9650
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware • 9608SIPCC • 9611SIPCC • 9621SIPCC • 9641SIPCC • 4620SIP with SIP firmware • SIP Softphone/Avaya one-X Desktop • Toshiba SP-1020A **Note:** Any model telephone that has SIP firmware and is being used for SIP networking must be administered as a 4620SIP telephone, 96xxSIP telephone, or 16CC SIP telephone. **Note:** Communication Manager does not support 1616SIP CC and 4620SIP CC telephones from Release 6.2 and later.	4620SIP
	Avaya one-X (tm) Deskphone 9620, 9630, 9630G 9640, 9640G with SIP firmware	96xx or 96xxSIP telephone
	9608 with SIP firmware	v
	9611 with SIP firmware	9611SIP

Table continues…

Telephone type	Model	Administer as
	9621 with SIP firmware	9621SIP
	9641 with SIP firmware	9641SIP
H.323 SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	Any NI-BRI (N1 and N2) telephone	NI-BRI
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		• XDID (use when Communication Manager later assigns a DID number to this station) • XDIDVIP (use when the administrator later assigns a DID number to this station) virtual (use to map this and other extensions to one physical telephone)
Key telephone system interface	—	K2500
ASAI	asai link computer telephony adjunct link	asai adjlk
AWOH	any digital set	same as “Multi-appearance Digital”
	CTI station	CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs

XMOBILE Type

The type of XMOBILE station. Available only if the **Type** field is XMOBILE and the **Mobility Trunk Group** field is administered.

Valid Entry	Usage
DECT	The DECT Access System or the AGCS (ROAMEO) IS-136 (TDMA cellular). This represent the ISDN-based DECT system.
EC500	Any public cellular networks.
IPDECT	The H.323 IP-based DECT system. This can either be the ASCOM or the DeTeWe product.
PHS	The DENSO 300M.

XOIP Endpoint type

Available only for 500, 2500, K2500 and CallrID station types.

Valid Entries	Usage
auto modem fax tty	Identifies the endpoint type of the analog station. Use this field for older or non-standard external equipment such as modems, fax, and TTY devices that are not easily recognized by VoIP resources within Communication Manager. By identifying this external equipment through administration, VoIP firmware can determine whether to immediately attempt to put a call in pass-through mode, or allow the system to handle it normally. Default is auto. **Note:** This field is intended for exception cases only. For the majority of stations, use the default setting of auto.

Chapter 118: Stations With Off-PBX Telephone Integration

Maps an office telephone to a cell phone through the Extension to Cellular feature. The office telephone can be a standard office number or an administration without hardware (AWOH) station.

Example command: `add off-pbx-telephone station-mapping`

Stations With Off-PBX Telephone Integration: page 1

Application

The type of off-PBX application that is associated with the office telephone. More than one application can be assigned to an office telephone.

Valid Entry	Usage
blank	Default is blank.
CSP	cell phone with Extension to Cellular provided by the cellular service provider
EC500	cell phone with Extension to Cellular
HEMU	Home Enterprise Mobility User
OPS	SIP enabled phone
PBFMC	Public Fixed Mobile Convergence
PVFMC	Private Fixed Mobile Convergence
SCCAN	wireless SIP phone and cell phone
SPFMC	Service Provider Fixed-Mobile Convergence
VEMU	Visited Enterprise Mobility User
VIEMU	Visited Initial Enterprise Mobility User
ONE-X	It is used by Avaya one-X® Client Enablement Services. You cannot enter any data for the ONE-X application types, but you can see this on the display command if it is active.

CC

The country code associated with the extension. Accepts up to three digits. Multiple entries that use the same telephone number must also have the same country code.

Country code changes made to existing stations or XMOBILE entries are applied to all instances of the telephone number.

SAFE (Self-Administered Feature Access Code for EC500) must not be used with an extension that has an administered country code. Origination mapping can occur with or without a country code. Default is blank.

Configuration Set

Valid Entry	Usage
1 to 99	The Configuration Set number that contains the required call treatment options for the Extension to Cellular station. The SCCAN application requires two different configuration sets selected for each station. The first set is the value for the WLAN followed by a slash. The second is the value for the cellular network.
blank	Enterprise Mobility User (EMU) and ONE-X application type display a blank entry.. The default setting is blank.

Dial Prefix

The dial prefix the system prepends to the off-switch telephone number before dialing the off-switch telephone. The system deletes the dial prefix when a user enters their cell phone number using the Self Administration Feature (SAFE) access code. The routing tables must be set properly so that the dial prefix 1 is not necessary for correct routing. Communication Manager inserts a plus (+) digit before initiating a call to a SIP endpoint or a cell phone.

- asterisk (*) or hash (#) must be in the first digit position
- 1 must be used if the telephone number is long-distance
- 011 must be used if the telephone number is international

Dual Mode

The **Dual Mode** field provides functionality for dual mode wireless phones that operate both in the public cellular network using GSM or CDMA and in a wireless network using SIP. The **Dual Mode** field indicates which application is part of a dual-mode pair. The dual Mode pair determines the pair of channels that the wireless phone is using, one channel over GSM/CDMA and the other channel

over SIP. The dual Mode pairing is required to transfer a call to the correct channel when the call is moving between GSM/CDMA and SIP networks.

Valid Entry	Usage
blank	The application is not part of a dual-mode pair.
DM1	The application belongs to a dual-mode pair. Applications of the same pair have the same value.
DM2	The application belongs to a dual-mode pair. Applications of the same pair have the same value.
DMX	Supported only for Private Fixed Mobile Convergence application for pairing with one-X application.

Note:

dm1 or dm2 can be assigned to any two of the Public Fixed Mobile Convergence (PBFMC), Private Fixed Mobile Convergence (PVFMC), and Service Provider Fixed-Mobile Convergence (SPFMC) applications. However, one of the applications must be PVFMC. The other application can be PBFMC or SPFMC.

Phone Number

The **Phone Number** field accepts:

- The telephone number of the off-switch telephone.
- The telephone number of an on-PBX extension for the one-X® application type.

The **Phone Number** field can be blank for:

- the first EC500, CSP, or PBFMC telephones you administer
- EC500, CSP, and PBFMC telephones that support self-administered Feature Access Code for EC500 (SAFE)

Station Extension

The number of the office telephone that is mapped to the cell telephone. Accepts up to thirteen digits. This number must be an administered extension. Default is blank.

Trunk Selection

Valid Entry	Usage
ars aar	The **Trunk Selection** field Indicates the trunk group selection method used for the outgoing call to Session Manager only if the **application** field is set to `ops`. If the **application** field is set to any other value, for

Valid Entry	Usage
trunk group number ext	example EC500, then the selected trunk will not be Session Manager. The selected trunk will be some other destination, for example, to a cell phone in the PSTN. In the half-call model, indicates a different trunk group than the trunk group that eventually connects the call to the telephone network. For the one-X® application, type: ars: for off-PBX telephones. ext: for on-PBX telephones.

Stations With Off-PBX Telephone Integration: page 2

Bridged Calls

Determines if bridged call appearances extend to the Extension to Cellular cell telephone.

Valid Entry	Usage
termination	Users can use their Extension to Cellular cell phone to only receive calls from the associated office telephone. Users cannot use the cell phone to originate calls from the associated office telephone. Calls originating from the cell phone independent of the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular. **Note:** The on-PBX extensions that the ONE-X application type enables are in termination mapping mode only.
origination	Users can originate Extension to Cellular cell phone calls only from the associated office telephone. Users cannot use the cell phone to receive calls from the associated office telephone.
both	Users can originate Extension to Cellular cell phone calls from the associated office telephone, as well as receive calls from the associated office telephone.
none	Users cannot originate or receive calls from the office telephone with the cell phone.

Call Limit

Use this field to set the maximum number of calls that can be made simultaneously by using an application, such as Extension to Cellular (EC500) or Off-Premise Station (OPS), on a station. You can enter a value from 1 to 10. For example, if you set the **Call Limit** field for an EC500 application to 2 and the **Call Limit** field for an OPS application to 4, the station can have up to two EC500 calls and up to four OPS calls.

Calls Allowed

Identifies the call filter type for an Extension to Cellular station. Determines the type of calls to the office telephone that a user can receive on an Extension to Cellular cell telephone.

Valid Entry	Usage
all	The cell telephone receives both internal and external calls. Default is all.
internal	The cell telephone receives only internal calls.
external	The cell telephone receives only external calls.
none	The cell telephone does not receive any calls made to the associated office telephone.

Location

Valid entry	Usage
1 to 2000	(Depending on your server configuration, see *Avaya Aura®Communication Manager System Capacities Table*, 03-300511.) The location value for each administered OPS, PBFMC, SPFMC or PVFMC application. Typically, set this to the same location number being used by the desk set. For more information on location, see *Avaya Aura® Communication Manager Feature Description*, 555-245-205.
blank	Trunk location is used for the outgoing calls from Off-PBX endpoints and location of station is used for incoming calls to Off-PBX endpoints. Blank is the default.

Related Links

Application on page 754
Loc Number on page 540

Mapping Mode

The mode of operation for the Extension to Cellular cell phone. These modes control the degree of integration between the cell phone and the office telephone. The modes are valid for Extension to Cellular calls only. For each office telephone, only one cell phone can be assigned as the origination mode. A cell phone cannot be assigned as either the origination or both mode more than once.

Valid Entry	Usage
termination	Users can use their Extension to Cellular cell phone to only receive calls from the associated office telephone. Users cannot use the cell phone to originate calls from the associated office telephone. Calls originating from the cell phone independent of

Table continues…

Valid Entry	Usage
	the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular. **Note:** The on-PBX extensions that the ONE-X application type enables are in termination mapping mode only.
origination	Users can originate Extension to Cellular cell phone calls only from the associated office telephone. Users cannot use the cell phone to receive calls from the associated office telephone.
both	Users can originate Extension to Cellular cell phone calls from the associated office telephone, as well as receive calls from the associated office telephone.
none	Users cannot originate or receive calls from the office telephone with the cell phone.

Stations With Off-PBX Telephone Integration: page 3

Share Level

Indicates the sharing level. Enter the value from 1-9.

Chapter 119: Survivable ARS Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Example command: `change survivable-ars-analysis`

Call Type

Valid Entry	Usage
emer npa hnpa intl iop locl natl op svc	The call type used for the dialed string.

Deny

Indicates whether or not the dialed string should be blocked. The system denies a dialed string that does not match an entered pattern.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, x and X.

Total Length

Valid Entry	Usage
0 to 28	The minimum number of digits required to validate the route. The minimum value when the dial string is populated is the length of the dialed string entry with a maximum value up to 28. Default is blank.

Trunk Grp No

The trunk group number that specifies the destination route for the dial plan analysis of this dialed string.

Chapter 120: Survivable Processor

Adds information specific to a Survivable Remote Server (Local Survivable Processor) or to connect certain adjuncts to a Survivable Remote or Survivable Core server (Enterprise Survivable Server). Before administering this screen, you must first assign node names for each Survivable Remote and Survivable Core media server on the IP Node Names screen.

While this screen is administered on the active main server, the information entered applies only to Survivable Remote and Survivable Core servers and does not apply to main servers. When translations are copied to a Survivable Remote or Survivable Core server, the Survivable Remote Server/Survivable Core Server replaces like translations for the main server with the overrides administered on the Survivable Processor screens. That is, use the Survivable Processor screen to administer overrides against adjunct links that have already been administered for the main servers. For more information about survivable core server, see *Avaya Aura® Communication Manager Survivable Options*, 03-603633.

Example command: `add survivable-processor n`, where *n* is the assigned node name.

Survivable processor: page 1

Cluster ID

Valid Entry	Usage
1 to 999	The Cluster ID for the survivable core server. The Cluster ID corresponds to the Module ID as found on the Server Role web page of the Survivable Core Server or Survivable Remote Server. **Note:** The **`statuslicense -v`** shell command on the survivable core server or the survivable remote server displays the Module ID.

Community

A community is a virtual group consisting of a survivable core server and one or more port networks. Assigning a survivable core server to a community associates the survivable core server with each IPSI in each port network for that community. Each IPSI is assigned to communities system-wide.

The association affects how the survivable core server is prioritized for the IPSI in that community, if the survivable core server is administered with a Local Preferred or Local Only preference. The Community number for an S8400 survivable core server must be set to 2 or greater and must be unique.

Related Links

System Parameters Port Networks on page 824

Community Size

Valid Entry	Usage
Sngl_PN	For a survivable core server, the value must be Sngl_PN.
all	Default is all.

Enable PE for H.248 Gateways

Enables or disables using the PE interface of the survivable core server for gateways.

Enable PE for H.323 Endpoints

Enables or disables using the PE interface of the survivable core server for H.323 devices such as telephones.

IP Address

The IP address that corresponds to the node name.

There are three IP addresses, one for each node name if the survivable processor is a duplicated survivable core.

Related Links

Name on page 451

Local Only

Enables or disables the survivable core server accepting a request for service from an IPSI. Can be enabled only if the IPSI is located in the same community as the survivable core server. Automatically enabled for survivable core servers and cannot be changed. Otherwise, this field defaults to disabled.

Related Links

System Preferred on page 765
Local Preferred on page 764

Local Preferred

Valid Entry	Usage
y	The survivable core server accepts a request for service from IPSIs co-located in the same geographical region, WAN segment, LAN segment, district, or business unit.
n	Default when the community size is set to Sngl_PN for an S8400 survivable core server and cannot be changed.

Related Links

Community Size on page 763
System Preferred on page 765
Local Only on page 763

Node Name

The previously-administered name used to identify the server. If the survivable processor is duplicated, there are three node names, one each for the duplicated server pair and one for the server that is active at a given point.

Priority Score

Valid Entry	Usage
1 to 100	The priority score for this survivable core server. Default is 1.

Processor Ethernet Network

Valid Entry	Usage
1 to 2000	The network region in which the PE interface of the Survivable Remote or Survivable Core server resides.

Server A — Server ID

Valid Entry	Usage
1 to 256	Server A ID corresponds to the Server ID configured on the survivable core server. The administration on the main server and the configuration on the survivable core server must match for the survivable core server to register to the main server.

Server B — Server ID

Valid Entry	Usage
1 to 256	For duplicated survivable core servers, displays the node name of Server B.

System Preferred

Valid Entry	Usage
y	Allows one survivable core server to replace the main server to keep as much of the system network intact as possible. Disables **Local Preferred** and **Local Only** and cannot be changed. This is the default except for an S8400 survivable core server.
n	The default for an S8400 survivable core server and cannot be changed.

Related Links

Local Only on page 763
Local Preferred on page 764

Type

Valid Entry	Usage
lsp simplex_ess duplex_ess	The survivable processor type.

Survivable Processor: page 2 (Processor Channels page)

Most fields on this page turn to display only when data for the processor channel is inherited from the main server.

Appl

Identifies the server application type or adjunct connection used on this channel.

Valid Entry	Usage
mis	Call Management System channel assignments.
ccr	Avaya IQ channel assignments.

Destination Node

The server or adjunct at the far end of this link consisting of an administered adjunct name, server name, far end IP address, or node name. Leave blank for services local to this server.

Destination Port

Valid Entry	Usage
0 5000 to 64500	The far-end port number of this link. An entry of 0 means any port can be used.

Enable

Specifies how data for this processor channel is transferred to the survivable processor.

Valid Entry	Usage
i(nherit)	This link is to be inherited by the Survivable Remote or Survivable Core server. The survivable processor inherits this processor channel just as it is administered on the main server. Used in the following situations: • The main server connects to the adjuncts using a CLAN and you want the Survivable Core server to use the same connectivity • The main server connects to the adjuncts using the PE interface of the main server, and you want the Survivable Remote or Survivable Core server to connect to the adjunct using its PE interface.
n(o)	This processor channel is disabled on the Survivable Remote or Survivable Core server. The survivable processor does not use this channel. This is the default.
o(overwrite)	The survivable processor overwrites the processor channel information sent in the file sync from the main server. With the entry o(overwrite), the administered adjunct attributes can be modified uniquely for each individual Survivable Remote or Survivable Core server.

Interface Channel

The channel number or the TCP/IP listen port channel to carry this processor (virtual) channel.

Valid Entry	Usage
5000 to 64500	For ethernet or ppp. For TCP/IP, interface channel numbers are in the range 5000 to 64500. The value 5001 is recommended for CMS, and 5003 is recommended for DCS.
0	Any port can be used.

Interface Link

Valid Entry	Usage
1 to 254	The physical link carrying this processor (virtual) channel.
pv4 or pv6	Communication Manager's Processor Ethernet interface used for adjunct connectivity.
blank	Not administered.

Mode

Valid Entry	Usage
c(lient)	The IP session is passive.
s(erver)	The IP session is active.

Proc Chan

Displays the processor channel number used for this link.

Related Links

Proc Chan on page 636

Session - Local/Remote

Valid Entry	Usage
1 to 384 blank	The Local and Remote Session numbers. For each connection, the Local Session number on the Avaya server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections. Local and Remote Session numbers must be consistent between endpoints.

Survivable Processor: page 3 (IP Services page)

Used when an AESVCS or a CDR connects to the Survivable Remote or Survivable Core server.

Enabled

Specifies how data for each specified service type is transferred to the survivable processor.

Valid Entry	Usage
i(nherit)	This link is to be inherited by the Survivable Remote or Survivable Core server. The survivable processor inherits this service type just as it is administered on the main server. Used in the following situations: • The main server connects to the adjuncts using a CLAN and you want the Survivable Core Server to use the same connectivity • The main server connects to the adjuncts using the main server's PE interface and you want the Survivable Remote or Survivable Core server to connect to the adjunct using its PE interface
n(o)	This IP services link is disabled on the Survivable Remote or Survivable Core server. This is the default.
o(verwrite)	Overwrites the processor channel information sent in the file sync from the main server. With the entry o(overwrite), the administered CDR or AE Services attributes can be modified uniquely for each individual Survivable Remote or Survivable Core server.

Local Node

Displays the previously-administered node name.

Related Links

Name on page 451

Local Port

The originating port number. For client applications such as Call Detail Recording (CDR), this field defaults to 0.

Remote Node

The name at the far end of the link for the CDR. Does not apply for AESVCS.

Remote Port

Valid Entry	Usage
5000 to 65500	The port number of the destination for CDR or AESVCS. This number must match the port administered on the CDR or AESVCS server.

Service Type

Valid Entry	Usage
CDR1 CDR2 AESVCS	The service provided.

Related Links

Service Type on page 474

Store to dsk

Enables or disables the storage of the CDR data on the local hard drive of the Survivable Remote or Survivable Core server. Pertains only to service types set to CDR1 or CDR2.

Survivable Processor: page 4 (IP Services — Session Layer Timers page)

Available only if CDR1 or CDR2 is administered, and if processor channel information is administered to be overwritten.

Related Links

Service Type on page 769
Enabled on page 768

Connectivity Time

Valid Entry	Usage
1 to 255	The amount of time that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up. Default is 60.

Packet Resp Timer

Valid Entry	Usage
1 to 255	The number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet. Default is 30.

Reliable Protocol

Enables or disables using a reliable protocol over this link. A reliable protocol should be used if the adjunct on the far end of the link supports it.

Service Type

Displays the previously-administered service type.

Related Links

Service Type on page 769

Session Connect Message Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager transmits a unit of protocol data before generating an error.

Chapter 121: System Capacity

Provides a status of administered capacity information and a snapshot status of system resources. For more information, see *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431. Detailed system capacity information can be found in *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Chapter 122: System Configuration

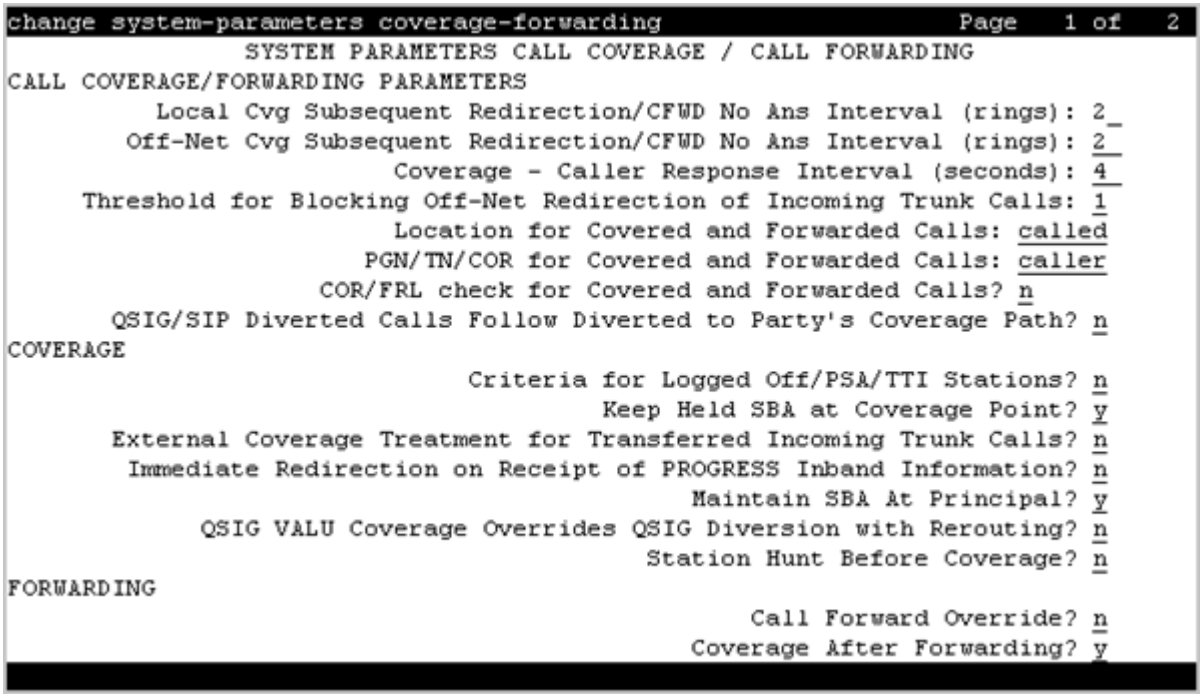
The System Configuration screen shows all the boards on the system that are available for connecting telephones. Used to view the board number, board type, circuit-pack type, and status of each board’s ports.

Chapter 123: System Parameters Call Coverage/Call Forwarding

Sets the system-wide parameters for call coverage and call forwarding.

Example command: `change system-parameters coverage-forwarding`

System parameters call coverage/ call forwarding: page 1



```
change system-parameters coverage-forwarding                    Page   1 of   2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING
CALL COVERAGE/FORWARDING PARAMETERS
          Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
        Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
                          Coverage - Caller Response Interval (seconds): 4
     Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1
                          Location for Covered and Forwarded Calls: called
                       PGN/TN/COR for Covered and Forwarded Calls: caller
                    COR/FRL check for Covered and Forwarded Calls? n
       QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path? n
COVERAGE
                                Criteria for Logged Off/PSA/TTI Stations? n
                                      Keep Held SBA at Coverage Point? y
       External Coverage Treatment for Transferred Incoming Trunk Calls? n
        Immediate Redirection on Receipt of PROGRESS Inband Information? n
                                                 Maintain SBA At Principal? y
             QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? n
                                           Station Hunt Before Coverage? n
FORWARDING
                                                  Call Forward Override? n
                                                Coverage After Forwarding? y
```


Comments on this document? infodev@avaya.com

CALL COVERAGE / FORWARDING PARAMETERS

Coverage - Caller Response Interval (seconds)

Valid Entry	Usage
0 to 10	The time in seconds an internal caller has before a call redirects to the called party's first coverage point.

COR/FRL check for Covered and Forwarded Calls

Valid Entry	Usage
y	Communication Manager checks the following: • COR of the calling and forwarded-to parties for local forwarding. • COR and FRL of the calling party and outgoing trunk for remote forwarding and remote coverage calls.
n	Communication Manager does not check the following: • COR of the calling and forwarded-to parties for local forwarding. • COR and FRL of the calling party and outgoing trunk for remote forwarding and remote coverage calls. This is the default value.

Diverted Party Identification

The **Diverted Party Identification** field is available only if the **QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path** field is enabled. The default setting is last-fwd.

The field settings do not work if the incoming call is over a non-QSIG or non-SIP trunk or the call is routed through a route-to vector step.

Valid entry	Usage
principal	To play voice mail greeting of the principal and save the voice message at the voicemail box of the principal.
last-fwd	To play voice mail greeting of the last-forwarded party and save the voice message at the voicemail box of the last-forwarded party. The default configuration.

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

Valid Entry	Usage
1 to 99	Specifies: • The number of rings applied at a local coverage point before a call redirects to the next coverage point • The number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated

Valid Entry	Usage
	Note: When ringing local destinations, such as an office environment, a short interval often is appropriate because the intended party either is near the telephone or not present. However, if the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Location for Covered and Forwarded Calls

Determines the location number used for coverage and forwarding.

Valid Entry	Usage
called	• If the called party is registered or in-service, coverage and forwarding use the called party's physical phone's location number. • If the called party is AWOH (x-port) or unregistered, coverage and forwarding use a location based on the type of set. - IP set: location 1 - DCP set: location 0 (all) • When the forwarding or coverage destination is to UDP instead of to an external destination starting with the ARS FAC, routing is always based on the caller's physical phone's location regardless of how this field is administered. This is the default.
caller	Coverage and forwarding use the caller's physical phone's location number.

Related Links

ARS FAC on page 538

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

Valid Entry	Usage
1 to 99	Specifies: • The number of rings applied at an off-net coverage point before a call is redirected to the next coverage point • The number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage. **Note:** When ringing local destinations, such as an office environment, a short interval often is appropriate because the intended party either is near the telephone or not present. However, if the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

PGN/TN/COR for Covered and Forwarded Calls

Valid Entry	Usage
called	Communication Manager checks for permissions (tenant number, partition group number, and COR) between the actual called party (forwarding party) and forwarded-to party. When the check is successful, the call is routed to the forwarded-to party. Permission check for COR is only done if the COR/FRL check for Covered and Forwarded Calls field is set to y.
caller	Communication Manager checks for permissions (tenant number, partition group number, and COR) between the calling party and forwarded-to party. When the check is successful, the call is routed to the forwarded-to party. Permission check for COR is only done if the COR/FRL check for Covered and Forwarded Calls field is set to y. This is the default value.

QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path

When a party with a defined coverage path does not answer the call, Communication Manager determines whether to continuously ring the call or redirect the call to a station or a messaging system. The calls follow the coverage path of the diverted-to party only if the calls route over QSIG and SIP trunks.

Valid entry	Usage
y	Communication Manager forwards the call to the coverage path of the last redirected party. The **Diverted Party Identification** field displays the default value as last-fwd.
n	The default configuration. The call continuously rings at the extension of the redirected party even if that extension has a coverage path. The call does not reach the messaging system of the last party in the coverage path.

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

Valid Entry	Usage
1 to 7	If the number of incoming trunk calls routes within the Call Forward timer, the block commences. Applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.
n (all)	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal can be redirected off-net.

COVERAGE

Criteria for Logged Off/PSA/TTI Stations

Enables or disables the call coverage criteria for logged-off IP/PSA/TTI stations. By default, the value is n.

Default treatment for logged off phones is a busy tone. If you set the **Criteria for Logged Off/PSA/TTI Stations** field to y, calls follow the coverage treatment as configured on the station screen. The call is redirected to coverage after the number of rings exceed the number specified in the **Number of Rings** field for **Criteria for Logged Off/PSA/TTI Stations**.

Related Links

Active on page 165

External Coverage Treatment for Transferred Incoming Trunk Calls

Enables or disables external coverage treatment for incoming trunk calls that redirect to coverage.

Immediate Redirection on Receipt of PROGRESS Inband Information

This field determines if a call is immediately redirected to coverage when an ISDN PROGRESS message is received. This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities.

However, the message might indicate that the cell telephone is not available to receive calls and should be redirected.

The system displays this field only if:

- The **Coverage of Calls Redirected Off-Net** field shows y.
- The Value-Added Avaya (VALU) feature is enabled for the system.

Valid entry	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point. Users in European countries following the ETSI standard and redirecting to GSM cellular telephones should enable the field.
n	Does not immediately redirect an off-net coverage/forwarded call to the next coverage point. Users in the United States should disable this field because PROGRESS messages with the **Progress Indicator** field set to inband information are sent for a variety of reasons that are not associated with unavailable cellular telephones.

Related Links

Cvg Of Calls Redirected Off-net on page 797
Value Added (VALU) on page 815

Keep Held SBA at Coverage Point

Determines how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid Entry	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but might enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

Maintain SBA At Principal

Using this, a user can maintain a simulated bridged appearance (SBA) when a call redirects to coverage.

Valid Entry	Usage
y	Maintains a simulated bridged appearance on the principal's telephone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	No SBA is maintained on the principal's telephone. DCS with rerouting is attempted, and if successful, the principal loses the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

Specifies whether or not, with both QSIG Diversion with Rerouting and QSIG VALU turned on, Coverage After Forwarding for the station works for calls that go to remote coverage.

Available only if **Basic Supplementary Services** and **Supplementary Services with Rerouting** are both enabled for the system.

Valid Entry	Usage
y	QSIG VALU call coverage takes precedence. If Coverage After Forwarding is enabled for a station, the call can receive coverage after rerouting.
n	With QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG server on which the forwarding destination resides. The forwarded call cannot return to coverage for the user who originally received the call.

Related Links

Basic Supplementary Services on page 814
Supplementary Services with Rerouting on page 814

Station Hunt Before Coverage

Determines whether or not a call to a busy station performs station hunting before going to coverage.

FORWARDING

Call Forward Override

Specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid Entry	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.

Table continues…

Valid Entry	Usage
n	Directs the system to forward calls to the forward-to station even when the calls are from the forwarded-to station.

Coverage After Forwarding

Determines whether or not an unanswered forwarded call is provided coverage treatment.

Valid Entry	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

System parameters call coverage/ call forwarding: page 2

```
change system-parameters coverage-forwarding                    Page   2 of   2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

                        Coverage Of Calls Redirected Off-Net Enabled? y
 Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                  Ignore Network Answer Supervision? n
                  Disable call classifier for CCRON over ISDN trunks? n
                   Disable call classifier for CCRON over SIP trunks? n

CHAINED CALL FORWARDING
                              Maximum Number Of Call Forwarding Hops: 3
                Station Coverage Path For Coverage After Forwarding: principal
```

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

Determines whether or not a simulated bridge appearance (SBA) is maintained on the principal when a call is directed to a final off-net coverage point. This field has no consequence when the off-

net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Available only if **Coverage of Calls Redirected Off-Net Enabled** is enabled.

Valid Entry	Usage
y	Maintains a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Drops the SBA on the principal's telephone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Related Links

Coverage Of Calls Redirected Off-Net Enabled on page 780

Coverage Of Calls Redirected Off-Net Enabled

Controls the Coverage of Calls Redirected Off-Net (CCRON) feature. Disables this feature if the demand on the call classifier port resources degrades other services provided by Communication Manager.

Available only if **Coverage of Calls Redirected Off-Net** is enabled for the system.

Valid Entry	Usage
y	Communication Manager monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Communication Manager does not monitor off-net coverage/ forwarded calls. No further coverage treatment is provided if the calls are unanswered.

Related Links

Cvg Of Calls Redirected Off-net on page 797

Disable call classifier for CCRON over ISDN trunks

Enables or disables the use of a call classifier on a CCRON call over ISDN facilities. When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and has precedence over ISDN trunk signalling. This field can direct Communication Manager to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid Entry	Usage
y	Disables the call classifier for CCRON calls over interworked trunk facilities.
n	Enables the call classifier for CCRON calls over interworked trunk facilities.

Disable call classifier for CCRON over SIP trunks

Valid Entry	Usage
y	Disables the call classifier for CCRON calls over interworked trunk facilities. Directs Communication Manager to dispense with the call classifier on interworked calls and rely on the SIP trunk signalling messages.
n	Enables the call classifier for CCRON calls over interworked trunk facilities.

Ignore Network Answer Supervision

Administers whether or not a call classifier for network answer supervision is used to determine when a call is answered. CCRON might use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that might indicate an answered state.

Available only if **Coverage of Calls Redirected Off-Net Enabled** is enabled.

Valid Entry	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered. Provides accurate answer supervision for tandem calls redirected to the public network.
n	Treat network answer supervision as a true answer. Preserves network answer supervision information.

Related Links

Coverage Of Calls Redirected Off-Net Enabled on page 780

CHAINED CALL FORWARDING

Maximum Number of Call Forwarding Hops

Available only if Chained Call Forwarding is enabled.

Valid Entry	Usage
3 to 10	The number of hops allowed in the forwarding chain.

Related Links

Chained Call Forwarding on page 331

Station Coverage Path For Coverage After Forwarding

This field specifies what coverage path the call must follow. The system displays the **Station Coverage Path For Coverage After Forwarding** field only when you enable the **Chained Call Forwarding** and the **Coverage After Forwarding** fields.

Valid entry	Usage
principal	The call follows the coverage path of the principal.
last-fwd	The call follows the coverage path of the last-forward-to party.

Related Links

Chained Call Forwarding on page 331
Coverage After Forwarding on page 779

Chapter 124: System Parameters Country Options

This screen implements parameters associated with certain international, including North American, call characteristics. This screen cannot be changed. Go to the Avaya Support website at http://support.avaya.com for current documentation, product notices, knowledge articles on how to modify any of the values here. This table shows the country codes that are used in Communication Manager. The Country Code is used by various fields and screens throughout the system.

Example command: `change system-parameters country-options`

Country options table

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
1	United States, Canada, Korea, India	300v peak to peak, < 200v peak to ground; < 70 Hz; < 5s on > 1s off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off
2	Australia, New Zealand	75 +/- 20 VRMS superimposed on 48 V dc at 14.5 to 55 Hz with cadence 400ms on, 200ms off, 400ms on, 2000ms off New Zealand: Ringing voltage at the customer's premises not less than 38 V rms (25Hz) on top of 50V d.c; 20 Hz; 400ms on, 200ms off, 400ms on, 2000ms off
3	Japan	75 VRMS(75-10VRMS <= x <= 75+8VRMS), 15-20 Hz and cadence of 1second on and 2 seconds off is required
4	Italy	20 to 50 Hz, 26 to 80 Volts rms superimposed on 48 V dc, 1 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off

Table continues…

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
5	Netherlands	25 Hz, 35 to 90 Volts rms superimposed on 66 V dc, 1 sec on, 4 sec off. Note that 50 Hz should be used, and another cadence can be 0.4 sec on, 0.2 sec off, 0.4 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
6	Singapore	75V at 24Hz with a cadence of 0.4 seconds on, 0.2 seconds off, 0.4 seconds on and 2.0 seconds off.
7	Mexico	25 Hz, 70 +/- 20 Vrms superimposed on 48Vdc Cadence 1 sec on, 4 sec off, flashhook is 100 ms
8	Belgium, Luxembourg, Korea	25 Hz, 25 to 75 Volts rms superimposed on 48 V dc, 1 sec on, 3 sec off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
9	Saudi Arabia	
10	United Kingdom	U.K.: 15 to 26.25 Hz, 25 to 100 Volts rms superimposed on 48 V dc, 0.35 on, 0.22 off then start in at any point in: 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off. Note 1: 48v DC can be present during the whole cadence or can be confined to silent periods. Note 2: Some exchanges provide a facility known as immediate ring; in this case an initial burst of ringing 20 msec to 1 sec in length immediately precedes switching to any point in the normal ringing cycle. Ireland: 25 Hz, 30 to 90 Volts rms superimposed on 50 V dc, 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off another possible cadence is 0.375 sec on, 0.250 sec off, 0.375 sec on, 2 sec off. ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
11	Spain	20 to 30 Hz, 35 to 75 Volts rms superimposed on 48 V dc, 1 to 1.5 sec on, 3 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
12	France	50 Hz, 28 to 90 Volts rms superimposed on 0.45 to 54 V dc, 1.5 sec on, 3.5 sec off

Table continues…

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
		ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
13	Germany	Germany: 25 Hz, 32 to 75 Volts rms superimposed on 0 to 85 V dc, 1 sec on, 4 sec off Austria: 40 to 55 Hz, 25 to 60 Volts rms superimposed on 20 to 60 V dc, 1 sec on, 5 sec off +/- 20% ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
14	Czech Republic, Slovakia	
15	Russia (CIS)	25+\-2 Hz, 95+\-5 Volts eff, local call cadence: first ring 0.3-4.5 sec then 1 second on 4 seconds Off toll automatic cadence 1 sec On 2 sec Off toll operator: manual sending
16	Argentina	25Hz; 75 Vrms superimposed on 48 Vdc; 1s on 4s off
17	Greece	
18	China	25Hz +/- 3Hz; 75 +/- 15 Vrms; Harmonic Distortion <= 10%; 1 sec ON, 4 secs OFF
19	Hong Kong	75 +/- 20 VRMS superimposed on -40 to -48 V dc at 25 Hz +/- 10% with cadence 0.4 s on, 0.2 s off, 0.4 s on, 3.0 s off
20	Thailand	
21	Macedonia	
22	Poland	
23	Brazil	25Hz +/-2.5Hz; minimum of 40 Vrms; 1s on, 4s off for equipment supporting up to six trunks only otherwise 25Hz +/-2.5Hz; minimum of 70+/-15 Vrms at a continuous emitting condition under no load, overlapping a DC level.
24	Nordic	Finland: 25 Hz, 35 to 75 Volts rms superimposed on 44 to 58 V dc, 1 sec on, 4 sec off 25 Hz, 40 to 120 Volts rms superimposed on 44 to 56 V dc, 0,75 on, 7,5 off +/- 20 % 25 Hz, 28 to 90 Volts rms superimposed on 24 to 60 V dc, 1 sec on, 4 sec off 25 and 50 Hz, 30 to 90 Volts rms superimposed on 33 to 60 V dc, 1 sec off, 5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off

Table continues…

Comments on this document? infodev@avaya.com

Code	Country	Ringing Signal Voltage, Frequency, and Cadence
25	South Africa	

Dial Tone Validation Timer (sec)

Available only when Tone Detection Mode is 4 or 5. Valid with TN420C or later Tone Detector circuit pack.

Valid Entry	Usage
0 to 6375	Displays number of milliseconds in increments of 25 that the dial tone validation routine uses to sample transmissions.

Related Links

Tone Detection Mode on page 788

Disconnect on No Answer by Call Type

Enables or disables the system from disconnecting calls that are not answered. Drops outgoing trunk calls, except DCS and AAR, that users leave unanswered too long.

Directory Search Sort Order

Available only for the Cyrillic or Ukrainian display character set.

Valid Entry	Usage
Cyrillic	Cyrillic Collation is used for integrated directory name search and result sorting. This is the default value.
Roman	Eurofont Latin Collation is used for directory name search and result sorting. The letters to be searched in the specified order for dial pad button presses are defined in the row for each key.

Related Links

Display Character Set on page 787

Display Character Set

Valid Entry	Usage
Cyrillic Katakana Roman Ukranian	The character set used for all non-native name values without an ASCII-only restriction.

Note:

Cyrillic, Roman, and Ukrainian map to the Eurofont character set. For Katakana, the Optrex font is used. If a Communication Manager server uses non-English in any name field, characters on a BRI station are displayed incorrectly.

Warning:

Changing the value in this field might cause some telephones to perform improperly, and can cause non-ASCII data in non-native names to display incorrectly on telephones. To correct this, non-native names of previously administered stations must be removed and re-administered using non-ASCII characters. This includes any display messages that have been administered.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

Enables or disables Busy Tone Disconnect. When enabled, Communication Manager recognizes a busy tone from the local telephone company central office as a disconnect signal.

Howler After Busy

Enables or disables howler tone when users leave their analog telephone off-hook too long.

Set Layer 1 timer T1 to 30 seconds

Specifies whether or not the Layer 1 timer is set to 30 seconds.

TONE DETECTION PARAMETERS

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored. (Valid with TN420C or later Tone Detector circuit pack.)

Valid Entry	Usage
short	5–30ms
long	20–40ms

Tone Detection Mode

Use this field to specify the type of tone-detection algorithm that Communication Manager must use for a TN420B or later tone-detection circuit pack.

Valid entry	Usage
1	The precise Italian tone-detection algorithm.
2	The precise Australian tone-detection algorithm.
3	The precise UK tone-detection algorithm.
4	The imprecise normal broadband-filter algorithm. This option is valid only for a TN420C or later tone-detection circuit pack.
5	The imprecise wide broadband-filter algorithm. This option is valid only for a TN420C or later tone-detection circuit pack.
6	The precise USA tone-detection algorithm.

Chapter 125: System Parameters Customer Options

Shows which optional features are enabled for the system, as determined by the installed license file. All fields on this screen are display only. For more information about disabling or enabling one of the features, go to Avaya Support website at http://support.avaya.com/.

Example command: `display system-parameters customer-options`

System parameters customer options: page 1

G3 Version

Identifies the version of Avaya Communication Manager being used.

Location

Indicates the location of this Avaya server or switch.

Valid Entry	Usage
1	Canada or the United States
2	Any other location. Allows the use of International Consolidation circuit packs and telephones.

Maximum Off-PBX Telephones - EC500

Stations that are administered for any Extension to Cellular (EC500/ CSP) application count against this limit.

The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.

- On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - OPS

Stations that are administered for any SIP Extension to Cellular/OPS application count against this limit. The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Note that legacy platforms do not support SIP trunks.
- On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones — PBFMC

Number of stations administered for Public Fixed-Mobile Convergence. Each station is allowed only one PBFMC application. The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.
- On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - PVFMC

Number of stations administered for Private Fixed-Mobile Convergence. Each station is allowed only one PVFMC application. The “license max” upper limit is:

- On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.
- On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - SCCAN

The “license max” value is defined as follows:

- SCCAN is only available on Linux systems. The upper limit is the maximum number of administrable stations.

Maximum Stations

Displays the maximum number of stations allowed in the system.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

Displays the platform being used.

Valid Entry	Usage
28	VCM
29	VCM_ESS
30	VCM_LSP

Platform Maximum Ports

Number of active ports.

Software Package

Indicates whether the software package license is Standard or Enterprise.

Used

The actual current usage as compared to the system maximum.

System parameters customer options: page 2

Maximum Administered Ad-hoc Video Conferencing Ports

Defines the number of ad-hoc ports allowed for the system; one for each simultaneous active conference port. The maximum number of ad-hoc video conferencing ports allowed is the sum of the maximum allowed IP trunks and the maximum allowed SIP trunks on your system.

Maximum Administered IP Trunks

Defines the maximum number of IP trunks administered.

Maximum Administered Remote Office Trunks

Defines the maximum number of IP endpoints based on the endpoint.

Maximum Administered SIP Trunks

Defines the maximum number of SIP trunks administered.

Maximum Concurrently Registered IP eCons

Specifies the maximum number of IP SoftConsoles that can be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that can be registered at one time.

Maximum G250/G350/G700 VAL Sources

Specifies the maximum number of VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Displays the number of DS1 circuit packs that can have echo cancellation.

Maximum Number of Expanded Meet-me Conference Ports

The maximum number of Expanded Meet-me Conference ports on the system.

Maximum TN2501 VAL Boards

The maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Maximum TN2602 Boards with 80 VoIP Channels

The total number of TN2602AP boards that can be administered with 80 VoIP channels.

Maximum TN2602 Boards with 320 VoIP Channels

The total number of TN2602AP boards that can be administered with 320 VoIP channels.

Maximum Video Capable IP Softphones

The maximum number of IP Softphones that are video-capable. The maximum number depends on the type of system.

Maximum Video Capable Stations

The maximum number of stations that are video-capable. The maximum number depends on the type of system.

Used

For each item with a capacity listed, the USED value is the actual number of units currently in use.

System parameters customer options: page 3

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Used to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1.

Analog Trunk Incoming Call ID

Used to collect and display the name and number of an incoming call on analog trunks.

Answer Supervision by Call Classifier

Indicates if the system can use a call classifier to do answer supervision. If the system has a call classification resource, the field is set to y. The field can be set to y even if the system has no call classifier circuit packs. This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a feature access code (FAC).

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into eight user groups within a single server running Avaya Communication Manager. Can establish individual routing treatment for each group.

ASAI Link Core Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by providing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

Note:

This field applies only to links administered as type asai.

ASAI Link Plus Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. If enabled, then the following ASAI capability groups are also enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Note:

This field applies only to links administered as type asai.

Asynch. Transfer Mode (ATM) PNC

PNC ATM PNC can be enabled only if:

- All prior fiber-link administration has been removed
- All “switch-node” and “dup-switch-node” carrier types have been removed.

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Enables circuit emulation service (CES).

Related Links

Carrier Medium on page 479

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

ATM WAN Spare Processor

Indicates whether or not an ATM WAN spare processor is part of the system. An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional.

Attendant Vectoring

Enables or disables Attendant Vectoring.

Audible Message Waiting

Enables or disables audible message waiting.

Authorization Codes

Enables or disables the use of Authorization Codes. Authorization Codes provide levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Enables or disables Centralized Attendant Service - Branch.

CAS Main

Enables or disables multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Avaya Communication Manager location. The main Avaya Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested

extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a telephone by using a feature access code (FAC).

Computer Telephony Adjunct Links

Provides linkage between Avaya Communication Manager and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

Note:

This field only applies to links administered as type adjlk.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of media servers or switches.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities.

Related Links

TSC Supplementary Service Protocol on page 688
Used for DCS on page 911
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801
Group Type on page 675

Digital Loss Plan Modification

Allows or disallows permission to customize the digital loss and digital tone plans.

DS1 MSP

Allows or disallows permission to administer values for the DS1 circuit pack without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

System parameters customer options: page 4

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya Communication Manager or dialed by users.

Enable ‘dadmin’ Login

Provides Avaya Business Partners the ability to install, administer, and maintain Avaya servers and switches. The dadmin login has access to all the same commands as other logins with the exception of **Go** and **WP**. **Go** is used for **go tcm** and **go debug** as well as **go server**. **WP** is for writing memory.

Enhanced Conferencing

Enables or disables the use of Meet-me Conference, Expanded Meet-me Conference, the No Hold Conference features, Selective Conference Party: Display, Drop, or Mute.

Enhanced EC500

Indicates if Extension to Cellular is enabled. EC500 refers to the Extension to Cellular feature.

Enterprise Survivable Server

Identifies the server is a Survivable Core Server (Enterprise Survivable Server).

ESS Administration

Indicates if administration of survivable core server (Enterprise Survivable Servers) is enabled.

Extended Cvg/Fwd Admin

Enables or disables Extended Coverage and Forwarding Administration.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. Go to the Avaya Support website at http://support.avaya.com/ for current documentation, product notices, knowledge articles on Enterprise Wide Licensing.

Five Port Networks Max Per MCC

Allows system administrator to create five port networks in a multi-carrier cabinet. Available only for duplex server Multi-Connect.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, and so on.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode).

Note:

Standard hospitality features are included in basic system software.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be enabled for IP telephones.

IP Trunks

Controls permission to administer H.323 trunks. Must be enabled for IP trunks.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager.

Related Links

Service/Feature on page 659
Facility Type on page 600
ISDN-PRI on page 801
TSC Supplementary Service Protocol on page 688
DCS with Rerouting on page 797
Used for DCS on page 911
ISDN-PRI on page 801
Group Type on page 675

ISDN Feature Plus

Provides ISDN Feature Plus signaling.

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be enabled for IP trunks.

Related Links

Service/Feature on page 659
Facility Type on page 600
ISDN-BRI Trunks on page 801
TSC Supplementary Service Protocol on page 688
DCS with Rerouting on page 797

Used for DCS on page 911
ISDN-BRI Trunks on page 801
Group Type on page 675

ISDN/SIP Network Call Redirection

Redirects an incoming ISDN/SIP call from a server running Avaya Communication Manager to another PSTN endpoint. It is used in call centers with Best Service Routing and Lookahead Interflow.

Local Survivable Processor

Indicates that the server is a Survivable Remote Server (Local Survivable Processor). The Survivable Remote Server is configured to provide standby call processing in case the primary media server is unavailable.

Malicious Call Trace

Provides the ability to retrieve certain information related to a malicious call.

Media Encryption Over IP

Enables or disables media encryption over IP network.

Mode Code for Centralized Voice Mail

Provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for multi-frequency signaling between Communication Manager and the local telephone company central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Multimedia IP SIP Trunking

Extends applicability of the H.323 video station licensing/control to all non-IP Softphones.

System parameters customer options: page 5

Multinational Locations

Provides the ability to use a single Enterprise Communication Server (ECS) with stations, port networks, remote offices, or gateways in multiple countries. Allows administration of location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Valid Entry	Usage
y	(Depending on your server configuration, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.) Allows numbering plans, time zone and daylight saving plans, digit analysis, companding, call progress tone generation, loss plan, and analog line board parameters that are specific for each cabinet, Gateway, Remote Office, network regions, signaling groups, and stations. See the Location sections in *Avaya Aura®Communication Manager Feature Description*, 555-245-205, for the other ways, and for a list of features that use location.
n	The location is fixed. The location is 1 for endpoints and all for digit analysis. This is the default value.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

PNC Duplication

Indicates whether or not Port Network Connectivity (PNC) Duplication can be enabled. This feature provides non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Valid Entry	Usage
y	The server is operating as a stand-alone Internal Communications Controller (ICC).
n	Traditional Avaya port networks are in use.

Posted Messages

Supports the ability for users to post messages, selected from among a set of as many as 30 (15 fixed, 15 administrable), and shown on display telephones.

Private Networking

Indicates upgrading of PNA or ETN software RTU purchased with earlier systems.

Processor and System MSP

Allows for maintenance of the processor and system circuit packs.

Processor Ethernet

Indicates if the Ethernet card resident in the processor cabinet is used by the Communication Manager Call Processing software in place of a C-LAN card or located in a port network. Only S8300D, S8300E S8510, and S8800 Media Servers display Processor Ethernet Interface. The Processor Ethernet interface is always enabled for duplex media servers.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

Allows the system to monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides the ability to use any data module as a secondary data module.

Station and Trunk MSP

Allows for maintenance of the station and trunk circuit packs.

Station as Virtual Extension

Allows multiple virtual extensions to be mapped to a single physical analog telephone. A specific ringing pattern can be administered for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

System Management Data Transfer

Indicates Communication Manager is accessible by network administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with X in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week to take advantage of lower calling rates during specific times.

Related Links

Time of Day Chart on page 124

TN2501 VAL Maximum Capacity

Allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs. This is the Enhanced offer.

Uniform Dialing Plan

Enables or disables three- to seven-digit Uniform Dial Plan (UDP) and one- to seven-digit steering. Also allows use of Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Enables the user to use user-defined entries for the incoming and outgoing ISDN calls independent of the NSF value. The Usage Allocation Enhancements feature lets the user administer the non-

unique Facility Type and Facility Coding combinations. With Usage Allocation Enhancements, the customer can reserve a minimum number of trunk members for the incoming and outgoing calls at all times.

Note:

The Usage Allocation Enhancements feature is helpful outside the US, where the default Network Facilities available in the system are not very useful.

Wideband Switching

Provides wideband data software for switching video or high-speed data. DSO channels can be aggregated up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for certain wireless applications.

System parameters customer options: page 6

ACD

Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Provides for hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be acceptable.

Business Advocate

Enables or disables Avaya Business Advocate. Business Advocate establishes different levels of service for different types of calls. For example, a company might decide that a premium customer gets faster service than other types of customers.

Call Center Release

The call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to voice response units (VRUs).

Dynamic Advocate

Enables or disables the Dynamic Advocate feature. While Business Advocate assigns reserve agents and sets overload thresholds to determine when those reserve agents get engaged, the Dynamic Advocate feature, also known as Dynamic Threshold Adjustment, takes this a step further. Dynamic Advocate automatically adjusts the thresholds as needed to help maintain defined service levels.

EAS-PHD

Increases the number of skills an agent can log in to from 4 to 20. Increases the number of agent skill preference levels from 2 to 16.

Expert Agent Selection (EAS)

Enables or disables skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See Multiple Call Handling.

Least Occupied Agent

Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature's access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's Vector Directory Number (VDN).

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing used to make vector routing decisions based on caller identity and the originating line.

Vectoring (Basic)

Provides basic call vectoring capability.

Related Links

Basic on page 89

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Vectoring (Best Service Routing)

Enables or disables the Best Service Routing feature. Through special vector commands, Best Service Routing allows the system to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (CINFO)

Enables or disables the Caller Information Forwarding (CINFO) feature that allows the collection of caller-entered digits (ced) and customer database provided digits (cdpd) for a call from the network.

Related Links

CINFO on page 89

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (Holidays)

Indicates if the Holiday Vectoring feature is enabled or disabled that simplifies vector writing for holidays.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

System parameters customer options: page 7

Logged-In ACD Agents

The total number of ACD agents that can be logged in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

The total number of Business Advocate Agents logged in simultaneously. The number of logged-in Business Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

The total number of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Puts call center statistics on agents, splits or skills, Vector Directory Numbers (VDNs), and trunk groups on telephone displays.

VuStats (G3V4 Enhanced)

Provides G3V4 VuStats enhancements including historical data and thresholds.

System parameters customer options: page 8 (ASAI FEATURES)

Agent States

Provides proprietary information used by Avaya applications. Go to the Avaya Support website at http://support.avaya.com/ for current documentation, product notices, knowledge articles on proprietary information used by Avaya applications.

Note:

This field applies only to links administered as type adjlk.

CTI Stations

Enables or disables any application using a link of Type ASAI that uses a CTI station to receive calls.

Phantom Calls

Indicates if phantom calls are enabled. This field only applies to links administered as type ASAI.

System parameters customer options: page 8 (QSIG OPTIONAL FEATURES)

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. Either ISDN-PRI or ISDN-BRI Trunks must be enabled for the system.

Related Links

ISDN-PRI on page 801
ISDN-BRI Trunks on page 801

Basic Supplementary Services

Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Either ISDN-PRI or ISDN-BRI Trunks must be enabled for the system.

Related Links

ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Centralized Attendant

Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled media server or switch in a network and a QSIG-enabled media server or switch:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.

- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Allows transfer directly into the voice mail box on the voice mail system when a QSIG link connects Avaya Communication Manager and the voice mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting.

System parameters customer options: page 9 (MAXIMUM IP REGISTRATIONS BY PRODUCT ID)

Limit

Valid Entry	Usage
1000 5000	Maximum number of IP registrations allowed, depending on server configuration.

Product ID

Identifies the product using the IP (internet protocol) registration.

These are just a few examples of valid Product IDs. The valid Product IDs for a system are controlled by the license file.

Product ID example	Description
IP_API_A	Application Enablement Services

Table continues…

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Product ID example	Description
IP_Agent	One-X agent
IP_eCons	Soft Console IP attendant
IP_Phone	IP telephones
IP_Soft	IP softphones
oneX_Comm	One-X Communicator

Rel

Valid Entry	Usage
*	IP endpoint can register with any release for the product ID.

Chapter 126: System parameters - Duplication

Use the System Parameters Duplication screen to enable PNC or IPSI duplication.

Example command: `change system-parameters duplication`

Enable Operation of IPSI Duplication

Enables or disables IPSI duplication.

Enable Operation of PNC Duplication

Enables or disables PNC duplication. Available only if PNC Duplication is enabled for the system.

Related Links

PNC Duplication on page 804

Chapter 127: System parameters maintenance

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431.

Chapter 128: System Parameters Media Gateway Automatic Recovery Rule

This screen defines rules for returning a fragmented network, where a number of Branch Gateways are being serviced by one or more Survivable Remote Server (Local Survivable Processors) automatically to the primary media server. The system displays a different warning message or time window grid depending on the option selected for the **Migrate H.248 MG to primary** field.

In the time window grid, an x or X can be specified for each hour. Leave blank to disable. This method helps with overlapping time issues between days of the week. There is no limit on intervals.

For more information on Auto Fallback for Branch Gateways, see *Administering Network Connectivity on Avaya Aura® Communication Manager*, 555-233-504.

Example command: `change system-parameters mg-recovery-rule` *n*, where *n* is the assigned recovery rule number.

Migrate H.248 MG to primary

Use this field to indicate auto-fallback preferences.

Valid entry	Usage
immediately	The system accepts the first gateway registration that comes from the gateway, regardless of call count or time of day. This is the default.
0-active calls	The system accepts the first gateway registration reporting a 0-active call count.
time-day-window	The system accepts a valid gateway registration message received during any part of the interval. When you select this option, the system displays a grid for defining required hours or days for the time window.
time-window-OR-0 - active-calls	The system accepts a valid gateway registration reporting a 0-active call count or a valid registration with any call count is received during the specified time-day intervals. When you select this option, the system displays a grid for defining required hours or days for the time window.

Minimum time of network stability

Valid Entry	Usage
3 to 15	Administers the time interval for stability in the H.248 link before auto-fallback is attempted. Default is 3.

Recovery Rule Number

The number of the recovery rule up to the server maximum.

Rule Name

The name for the recovery rule in alpha-numeric characters. The recovery rule is an aid for associating rules with gateways.

Chapter 129: System parameters OCM call classification

Administers the country tone characteristics for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. This screen defines the busy tone and cadence and can be administered with up to four on and off steps, which is four valid cycles to determine busy tone.

Use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is BTD signal.

Available only if **Global Call Classification** is enabled for the system, or when **Enable Busy Tone Disconnect for Analog loop-start Trunks** is enabled for the system.

Example command: `change system-parameters ocm-call-classification`

Related Links

Global Call Classification on page 800
Enable Busy Tone Disconnect for Analog Loop-start Trunks on page 787

System parameters OCM call classification: page 1

Cadence Classification After Answer

Valid Entry	Usage
y	Communication Manager puts call classifiers into modes that listen for call progress tones while listening the voice, with or without Answering Machine Detection (AMD).
n	No-cadence call classification modes detect live voice with or without AMD whenever Communication Manager receives a connect or answer supervision message from an outbound trunk. These modes do not detect any call progress tone cadence.

Global Classifier Adjustment (dB)

Valid Entry	Usage
0 to 15	Specifies the dB loss adjustment. where 0 is the least adjustment, and 15 is the most adjustment.

USA Default Algorithm

Enables or disables the use of the default United States tone detection.

USA SIT Algorithm

Valid Entry	Usage
y	Uses the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name intercept as if they were SIT VACANT, and treats tones with the administered tone name information as if they were SIT UNKNOWN.

System parameters OCM call classification: page 2

Cadence Step

Identifies the number of each tone cadence step and indicates whether the tone is on or off during this cadence step.

Duration Maximum

Valid Entry	Usage
75 to 6375	The upper limit in milliseconds of the tone duration in increments of 25.

Duration Minimum

Valid Entry	Usage
75 to 6375	The lower limit in milliseconds (msec) of the tone duration in increments of 25.

Instance

Valid Entry	Usage
1 to 8	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid Entry	Usage
y	Indicates a continuous tone.
n	Indicates a non-continuous tone.

Tone Name

Valid Entry	Usage
busy information intercept reorder ringback	The name of the tone. Busy is required when **Busy Tone Disconnect for Analog Loop-start Trunks** is enabled or **Global Call Classification** is disabled for the system. Required for tone definition outside of the U.S. and Canada.

Related Links

Global Call Classification on page 800
Enable Busy Tone Disconnect for Analog Loop-start Trunks on page 787

Chapter 130: System Parameters Port Networks

These screens assign port networks to communities and specify recovery rules for port networks to return to the main server.

Example command: `change system-parameters port-networks`

System parameters port networks: page 1

Community

Valid Entry	Usage
1 to 64	The Network Community number associated with this port network. **Note:** If the port network is administered in the system, the default community is 1 and administrable with a value between 1 and 64. If the port network is not administered in the system, the community value is 1 and not administrable.

PN

Valid Entry	Usage
1 to 64	Displays the port network.

System parameters port networks: page 2

Auto Return

The Auto Return functionality is used to schedule a day and time for all port networks to return to the control of the main server after a failover occurs. The schedule can be set up to seven days prior the its activation.

Valid Entry	Usage
y	Enables the Auto Return feature. The port networks can automatically return to the main server after the value set in the **IPSI Connection up time** expires.
n	Disables the Auto Return feature. The port networks cannot automatically return to the control of the main server.
s(cheduled)	Enables the Auto Return feature. A day and time also needs to be administered to schedule a day and time to return the port networks to the control of the main server.

Day

Valid Entry	Usage
Monday through Sunday	The day of the week the port networks return to the control of the main server if Auto Return is administered on a schedule.

IPSI Connection Up Time (min)

Valid Entry	Usage
3 to 120	The number of minutes that the IP Server Interface (IPSI) waits to return to the main server after communication with the main server is restored.

No Service Time Out Interval

Valid Entry	Usage
3 to 15	The time in minutes that the IP Server Interfaces (IPSIs) wait before requesting service from the highest survivable core server on its priority list. Default is 5 minutes.

PN Cold Reset Delay Timer (sec)

Valid Entry	Usage
60 to 120	Time in seconds before a PN cold reset occurs. This value is retained after an upgrade event. Default is 60 seconds.

Time

Valid Entry	Usage
00:00 to 23:59	The time of day the port networks return to the control of the main server if Auto Return is administered on a schedule. This field uses a 24-hour military format.

Chapter 131: System Parameters - SCCAN

Example command: `change system-parameters sccan`

Announcement

The extension of the announcement played during call hand-in or handout.

H1 Handover

The primary handover extension called to facilitate handover of a cellular call to the WAN or WLAN. Depending on whether the user is entering or exiting the Enterprise space, Communication Manager replaces the active call with the new call made using the hand-off H1 or H2 number.

H2 Handover

A secondary handover extension used when no acknowledgement is received from the H1 Handover number.

MM (WSM) Route Pattern

A route pattern number that is SCCAN-enabled. Partition route pattern indexes, RHNPA indexes, deny, or nodes are not supported. If this field is left blank, the feature is turned off. Blank is the default.

Special Digit Conversion

Valid Entry	Usage
y	Allows a user to call a cellular telephone number and get the same treatment as calling an extension that is running Communication Manager. ARS checks the dialed string to determine if the dialed string is a SCCAN telephone number. If the number is a SCCAN telephone number, the cellular telephone number is replaced with the extension number that the cellular telephone is mapped to.
n	The feature is turned off. This is the default.

Chapter 132: System Parameters Special Applications

Use this screen to enable the Special Application features. A Special Application feature is a custom feature requested by the customer for a specific need. Special Application features are included in the Communication Manager software base, however additional administration is required. For more information about the Special Application features, see *Avaya Aura® Communication Manager Special Application Features*.

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Chapter 133: Telecommuting Access

Administers the extension which allows remote users to use the Telecommuting Access feature.

Example command: `change telecommuting-access`

Telecommuting Access Extension

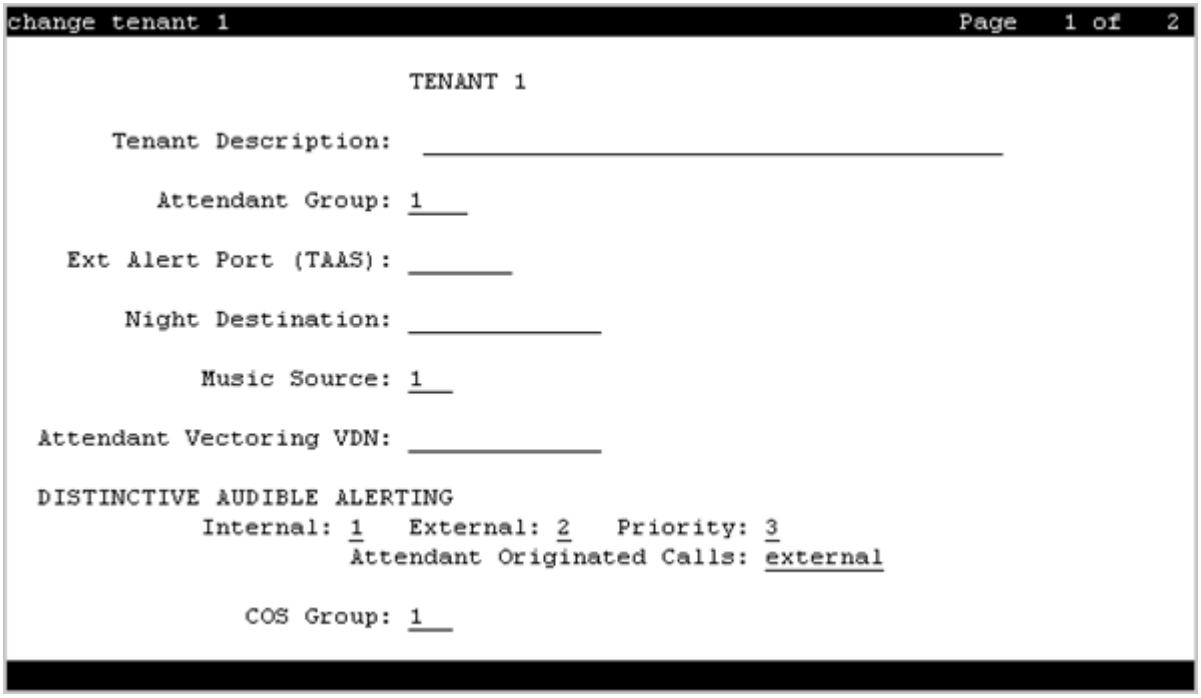
An extension that allows only remote access to the Telecommuting Access feature. This extension must consist of one to 13 digits that conforms to the system dial plan and is not assigned to any other system object.

Chapter 134: Tenant

This screen defines tenants to the system. If your server running Communication Manager uses tenant partitioning, see Tenant Partitioning in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Example command: `change tenant n`, where *n* is the tenant partitioning number.

Tenant: page 1



Attendant Group

Valid Entry	Usage
digits	The attendant group assigned to the tenant. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Attendant Vectoring VDN

Assigns the VDN extension for Attendant Vectoring to a console. Available only if Attendant Vectoring is enabled for the system and Tenant Partitioning is disabled.

Related Links

Attendant Vectoring on page 796
Tenant Partitioning on page 806

COS Group

Valid Entry	Usage
digits	The Class of Service (COS) group assigned to the tenant. Available only if Tenant Partitioning is enabled for the system. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Related Links

Tenant Partitioning on page 806

Ext Alert Port (TAAS)

The Trunk Answer Any Station (TAAS) alert port assigned to this tenant, if any. The port type and the object type must be consistent. The port can be assigned to only one tenant.

Valid Entry	Usage
A valid port address or X 1 to 64	First and second characters are the cabinet number
A to E	Third character is the carrier
0 to 20	Fourth and fifth character are the slot number
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number
1 to 250	Gateway

Ext Alert (TAAS) Extension

The extension of the external alert TAAS extension. A system installer can then use the Terminal Translation Initialization (TTI) feature from a telephone plugged into any port to assign this extension number to that port. Doing so makes that port the external alert TAAS port.

Available only if **Ext Alert Port (TAAS)** is administered without hardware.

Related Links

Ext Alert Port (TAAS) on page 832

Music Source

Valid Entry	Usage
digits	The music or tone source for this tenant partition. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

Night Destination

The night service station extension, if night service is wanted for this tenant.

Tenant

Displays the tenant number.

Tenant Description

A description of the tenant that consists of up to 40 alpha-numeric characters.

DISTINCTIVE AUDIBLE ALERTING

Available only if **Tenant Partitioning** is enabled for the system.

Attendant Originated Calls

Indicates which type of ringing applies to attendant-originated calls.

Valid Entry	Usage
internal	Internal ringing applies to attendant-originated calls.
external	External ringing applies to attendant-originated calls. Default is external.
priority	Priority ringing applies to attendant-originated calls.

Distinctive Audible Alerting (Internal, External, Priority)

The number of rings for Internal, External, and Priority calls. For virtual stations, this applies to the mapped-to physical telephone. This is also known as Distinctive Ringing. Defaults are as follows:

1. Internal calls
2. External and attendant calls
3. Priority calls

Note:

SIP messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid Entry	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

Related Links

Distinctive Audible Alerting (Internal, External, Priority) on page 296

Tenant: page 2

This screen is used to enable additional tenant-to-tenant calling permissions.

```
change tenant 1                                              Page   2 of   2

                                  TENANT 1

   CALLING PERMISSION (Enter "y" to grant permission to call specified Tenant)


  1? y    11? y    21? y    31? y    41? y    51? y    61? y    71? y    81? y    91? y
  2? y    12? y    22? y    32? y    42? y    52? y    62? y    72? y    82? y    92? y
  3? y    13? y    23? y    33? y    43? y    53? y    63? y    73? y    83? y    93? y
  4? y    14? y    24? y    34? y    44? y    54? y    64? y    74? y    84? y    94? y
  5? y    15? y    25? y    35? y    45? y    55? y    65? y    75? y    85? y    95? y
  6? y    16? y    26? y    36? y    46? y    56? y    66? y    76? y    86? y    96? y
  7? y    17? y    27? y    37? y    47? y    57? y    67? y    77? y    87? y    97? y
  8? y    18? y    28? y    38? y    48? y    58? y    68? y    78? y    88? y    98? y
  9? y    19? y    29? y    39? y    49? y    59? y    69? y    79? y    89? y    99? y
 10? y    20? y    30? y    40? y    50? y    60? y    70? y    80? y    90? y   100? y
```

Calling permissions

Establishes or blocks calling permission between the tenant being administered and any other tenant. The system default allows each tenant to call only itself and Tenant 1.

Tenant

Displays the tenant number.

Chapter 135: Terminal Parameters

Administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, 8434D, and 2420/2410 telephones. Only authorized Avaya personnel can administer this screen.

Note:

With the Multinational Locations feature enabled, terminal parameters can be administered per location, rather than system-wide.

Example command: `change terminal-parameters`

Base Parameter Set

Determines which default set of telephone options and levels will be used. This field corresponds to the country codes.

Related Links

Country options table on page 783

Customize Parameters

Allows or disallows permission for the administrator to change one or more of the default parameters.

ADJUNCT LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a user presses a button.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

OPTIONS

Display Mode

Determines how the # and ~ characters appear on the telephone's display.

Valid Entry	Usage
1	The # and ~ do not change.
2	The telephone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust.

Handset Expander Enabled

Determines whether or not the telephone reduces background noise on the handset.

Volume for DCP Types

Determines what volume adjustments are retained. Allows the DCP telephone volume to be adjusted while the call is in-progress.

Valid Entry	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

Determines what volume adjustments are retained. Allows the IP telephone volume to be adjusted while the call is in progress.

Note:

Avaya recommends that values for PRIMARY LEVELS or BUILT-IN SPEAKER LEVELS are not changed.

Valid Entry	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained
retain handset and speaker between calls	The adjusted handset and speaker settings are retained
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained

Touch Tone Transmit (dB)

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

PRIMARY LEVELS

The system displays the default setting from the Base Parameter Set for all fields. Also, these fields all require the same input; valid entries are from -44.0 db through +14.0 db in 0.5 increments. For example, -44.0, -43.5, -43.0 and so on.

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a user presses a button.

Touch Tone Transmit (dB)

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

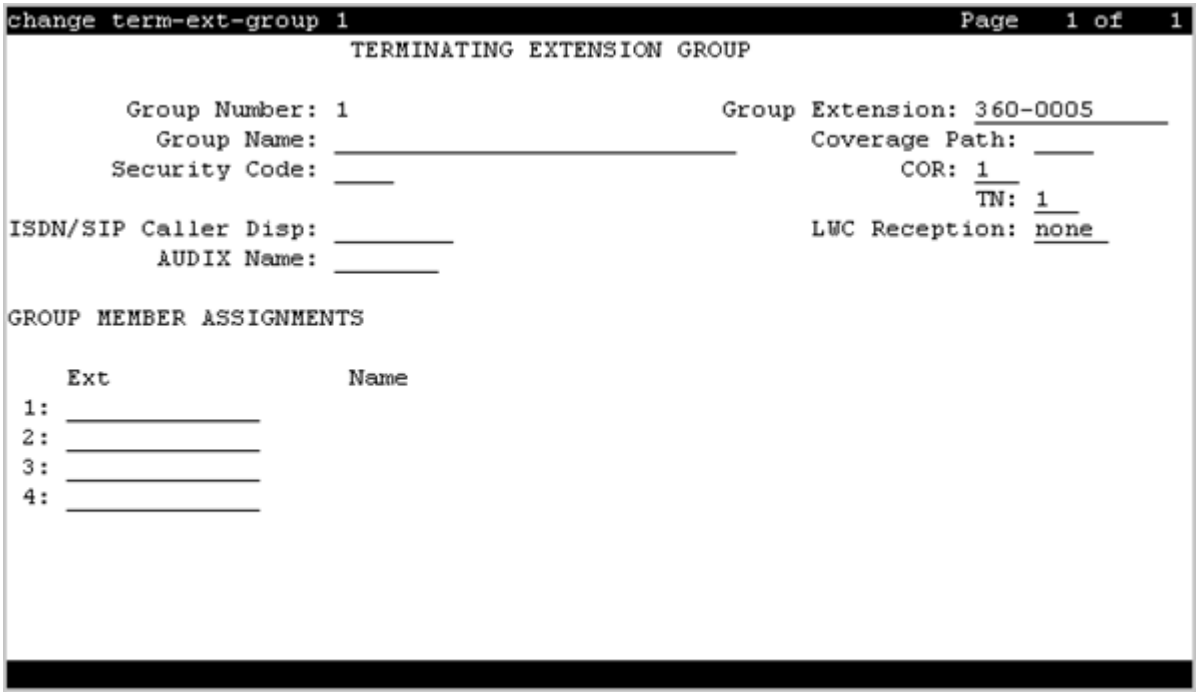
Chapter 136: Terminating Extension Group

Defines a Terminating Extension Group (TEG). Any telephone can be assigned as a TEG member; however, only a multi-appearance telephone can be assigned a TEG button with associated status lamp. The TEG button allows the telephone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member's COR. The members could all be termination restricted but still receive calls if the group is not restricted.

An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Example command: `change term-ext-group` *n*, where *n* is the assigned group number.



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AUDIX Name

The name of the AUDIX machine. Must be the same name as the IP Node name and administered *after* the IP Node is configured.

Related Links

IP Node Names on page 451
IP Node Names on page 451

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number that reflects the required restrictions.

Coverage Path

The call coverage path number for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid Entry	Usage
digits	Path number. For the maximum allowed entry, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.
t1 to t999	Time of day table
blank	Not administered

Group Extension

The extension of the terminating extension group. The extension can consist of one to seven digits and cannot be a Vector Directory Number (VDN). This field cannot be left blank.

Group Name

The name used to identify the terminating extension group.

Group Number

Displays the terminating extension group number.

ISDN Caller Disp

Specifies whether the TEG group name or member name (member of TEG where call terminated) is sent to the originating user. Required if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
grp-name	TEG group name is sent.
mbr-name	Member name is sent.
blank	Not administered.

Related Links

ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related Links

AUDIX Name on page 389

Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

GROUP MEMBER ASSIGNMENTS

Ext

The extension number of one to seven digits assigned to a group number. The extension cannot be a Vector Directory Number (VDN).

Name

Displays the name assigned to the group number.

Chapter 137: TFTP Server

The Trivial File Transfer Protocol screen allows specification of the TFTP server that Avaya Communication Manager uses to get download files.

Example command: `change tftp-server`

Filename in Memory

Displays the name of the file currently in Communication Manager memory.

File Size

Displays the number of bytes transferred.

File Status

Displays the download status as Download In Progress, Download Failed, File Not Found, or Download Completed.

File to Retrieve

The name of the file to retrieve using up to 32 alphanumeric, case sensitive, characters for identification.

Local Node Name

A previously-administered local node name. The node must be assigned to a CLAN IP interface or procr (processor CLAN).

Valid Entry	Usage
1 to 15	The node name.
procr	Processor CLAN for S8300 Server

Related Links

Name on page 451

Station Type

Displays the information related to the firmware, font, language, boot, and Digital Signal Processor (DSP) when a firmware, language file, font, DSP, or boot loader file download successfully.

TFTP Server Node Name

A previously-administered TFTP server node name from 1 to 15 characters.

Related Links

Name on page 451

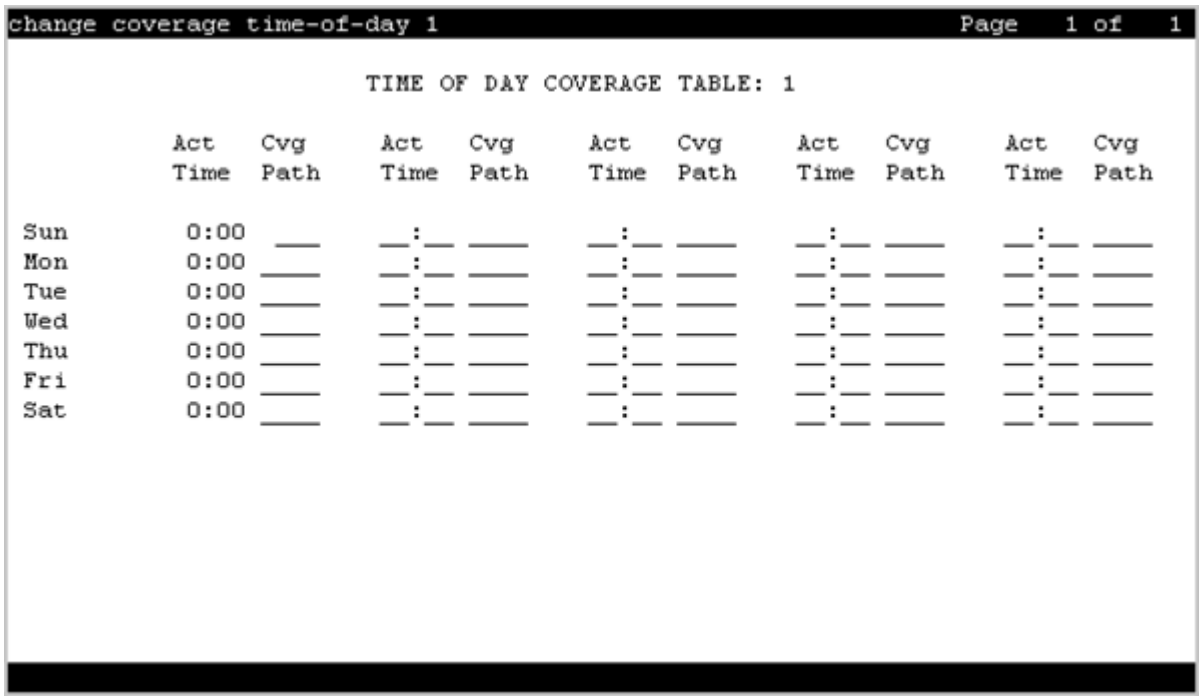
TFTP Server Port

Valid Entry	Usage
1 to 64500	The number for the remote TCP port.

Chapter 138: Time of Day Coverage Table

This screen allows administration of up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Example command: `change coverage time-of-day n`, where *n* is the assigned time of day coverage table.



Act Time

Valid Entry	Usage
00:01– 23:59	Specifies the activation time of the associated coverage path. Information must be entered in 24-hour time format. If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Valid Entry	Usage
1 to 9999 blank	The coverage path number.

Time of Day Coverage Table

Displays the Time of Day Coverage Table number.

Chapter 139: Time of Day Routing Plan

This screen is used to set up Time of Day Routing Plans.

- AAR and ARS calls can be routed based on the time of day each call is made.
- Up to eight Time of Day Routing Plans can be designed, each scheduled to change up to six times a day for each day in the week.
- The Time of Day Routing Plan PGN# is matched with the **PGN#** field on the Partition Routing Table for the correct route pattern.

Available only if **Automatic Route Selection (ARS)** or **Private Networking, AAR/ARS Partitioning**, and **Time of Day Routing** are enabled for the system.

Example command: `change time-of-day` *n*, where *n* is the time of day routing plans.

Related Links

PGN 1 (through PGN 8) on page 608
ARS on page 794
Private Networking on page 804
Time of Day Routing on page 806

Act Time

Valid Entry	Usage
00:00 to 23:59	Specifies the time of day the route pattern, identified by PGN#, begins. Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

The route pattern for the activation time listed. The PGN must match a previously-administered PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day.

Valid Entry	Usage
1 to 8	The route pattern range when SA9050 is inactive.
1 to 32	The route pattern range when SA9050 is active. The range depends on the type of CM template. The default value is 8. (For more information on how SA9050 changes capacities, see Avaya Aura® Communication Manager System Capacities Table, 03-300511.)

Time of Day Routing Plan

Displays the Time of Day Routing Plan number.

Chapter 140: Time of Day Station Lock Table

This screen administers the ability to lock stations automatically by a time of day schedule.

Example command: `change tod-station-lock n`, where *n* is the assigned time of day station lock table number.

Begin Time

Valid Entry	Usage
00:00 to 23:59	When the time of day station lock interval begins for each day of the week. Time is represented using a 24-hour clock.

End Time

Valid Entry	Usage
00:00 to 23:59	When the time of day station lock interval ends for each day of the week. Time is represented using a 24-hour clock.

INTERVAL 1, 2, 3

Three separate time of day station lock intervals can be administered. The system imposes validation of overlapping intervals or invalid blank entries.

Manual Unlock allowed

Indicates whether or not the user can manually unlock the TOD-locked station using either a **sta-lock** button or a Feature Access Code followed by an SSC.

Table Active

Activates or deactivates time of day lock for all stations associated with this table.

Chapter 141: Toll Analysis

This screen associates dialed strings to the system's Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. Users can be forced to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize system security, toll calling areas must be restricted as much as possible through the use of the **RCL** (Restricted Call List) and **Toll List** fields on this screen.

Note:

The Toll List field on this screen does not interact with or relate to the ARS Toll Table.

Example command: `change toll n`, where *n* is the dialed digits.

CDR FEAC

Valid Entry	Usage
x	Requires an account code from a call whose facility COR requires a Forced Entry of Account Code.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Location

Valid Entry	Usage
1 to 250 or 1 to 2000 Depending on your server configuration, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.	The location of the endpoint that is dialing the digits. Available only if **ARS** and **Multiple Locations** are enabled for the system. See the Location sections in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205, for the other ways to assign a location, and for a list of features that use location.
all	Indicates that this Toll Analysis Table is the default for all locations.

Related Links

ARS on page 794
Multiple Locations on page 804

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*, 03-300511.

RCL

Valid Entry	Usage
x	Assigns the Dialed String to the Restricted Call List (RCL). All entries marked with x and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR.

Toll List

Valid Entry	Usage
x	Assigns the Dialed String to the Toll List.

Dialed String	Minimum	Maximum	Toll List
0	1	23	X
1	4	23	X
20	10	10	X
21	10	10	X
30	10	10	X
31	10	10	X
40	10	10	X
41	10	10	X
50	10	10	X
51	10	10	X
60	10	10	X
70	10	10	X
71	10	10	X
80	10	10	X
81	10	10	X
90	10	10	X
91	10	10	X

Unrestricted Call List

Valid Entry	Usage
x	Assigns the dialed string to one of the system's Unrestricted Call Lists (UCL).

Chapter 142: Tone Generation

This screen administers the tone characteristics that parties on a call hear under various circumstances.

Note:

With the Multinational Locations feature enabled, tone generation can be administered per location, rather than system-wide.

Tone Generation: page 1

440Hz PBX-dial Tone

Valid Entry	Usage
y	The switch primary dial tone is changed to a continuous 440Hz-17 tone.
n	A customized tone defaults to the **Base Tone Generation Set**.

Related Links

Base Tone Generator Set on page 857

440Hz Secondary-dial Tone

Valid Entry	Usage
y	The secondary local telephone company central office dial tone is changed to a continuous 440Hz-17 tone.
n	A customized tone defaults to the **Base Tone Generation Set**.

Related Links

Base Tone Generator Set on page 857

Base Tone Generator Set

Valid Entry	Usage
1 to 50	The country code that identifies the base tone generation set to be used.

Related Links

Country options table on page 783

Tone generation: page 2 (TONE GENERATION CUSTOMIZED TONES page)

Cadence Step

Displays the number of each tone cadence step.

Duration (msec)

Valid Entry	Usage
50 to 12750	The duration of this step in the tone sequence. Values are in increments of 50.

Step

The number of the cadence step for this `goto` command.

Tone (Frequency/Level)

Valid Entry	Usage
silence	No tone. A final step of silence with an infinite duration will be added internally to any tone sequence that does not end in a goto.
goto	Repeats all or part of the sequence, beginning at the specified cadence step.
350/-17.25 350+425/-4.0	Specifies the frequency and level of the tone.

Table continues…

Valid Entry	Usage
350+440/-13.75	
375+425/-15.0	
404/-11.0	
404/-16.0	
404+425/-11.0	
404+450/-11.0	
425/-4.0	
425/-11.0	
425/-17.25	
440/-17.25	
440+480/-19.0	
450/-10	
480/-17.25	
480+620/-24.0	
525/-11.0	
620/-17.25	
697/-8.5	
770/-8.5	
852/-8.5	
941/-8.5	
1000/0.0	
1000/+3.0	
1004/0.0	
1004/-16.0	
1209/-7.5	
1336/-7.5	
1400/-11.0	
1477/-7.5	
1633/-7.5	
2025/-12.1	
2100/-12.1	
2225/-12.1	
2804/-16.0	

Tone Name

Valid Entry	Usage
1-call-wait 2-call-wait 3-call-wait busy busy-verify call-wait-ringback conference confirmation disable-dial hold hold-recall immed-ringback intercept intrusion mntr/rec-warning PBX-dial recall-dial recall-dont-ans redirect reorder rep-confirmation reset-shift ringback secondary-dial special-dial whisper-page zip	Indicates which of the individually administrable tones is modified.

Chapter 143: Trunk Group

Trunk Group

Sets basic characteristics for every type of trunk group and assigns ports to the group. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in **Group Type** might significantly change the content and appearance of the Trunk Group screen.

Note:

For descriptions of the screens and fields that are unique to ISDN trunks, see *ISDN Trunk Group*.

Trunk Group: page 1

```
change trunk-group 1                                          Page   1 of  22
                                TRUNK GROUP

Group Number: 1                      Group Type: co           CDR Reports: y
  Group Name: Trunk Group 1                  COR: 1      TN: 1      TAC:
   Direction: two-way          Outgoing Display? n
 Dial Access? n                Busy Threshold: 255  Night Service:
Queue Length: 0    Country: 1                 Incoming Destination:
   Comm Type: voice                    Auth Code? n   Digit Absorption List:
    Prefix-1? y                     Trunk Flash? n         Toll Restricted? y

         Trunk Type:
```

Analog Gain

Reduces the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume.

Available only if the country code is 15 and the **Trunk Type (in/out)** value is 2-wire-ac, 2-wire-dc, or 3-wire.

Valid Entry	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. This setting should not be changed unless the trunk group sound quality is unacceptable.

Related Links

Country on page 864
Trunk Type (in/out) on page 876

Auth Code

The **Auth Code** field is available only for incoming or two-way trunk groups if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk.

The following table provides the situations when the caller must enter an auth code.

Called party	Calling party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the calling station is less than the FRL of the outgoing route pattern
Incoming trunk	Station	No	the **Auth Code** field is enabled on the incoming trunk group
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern or the **Auth Code** field is enabled on the incoming trunk group

Note:

The **Auth Code** field is unavailable if:

- the **Group Type** field is tandem
- the **Group Type** field is ISDN and the **Service Type** field is tandem.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

Related Links

Direction on page 481
Authorization Codes on page 796

BCC

Generalized Route Selection uses the Bearer Capability Class (BCC) to select the appropriate facilities for routing voice and data calls. Far-end tandem servers/switches use the BCC to select outgoing routing facilities with equivalent BCC classes. The **BCC** entry is used to select the appropriate facilities for incoming ISDN calls. Communication Manager compares the **BCC** entry with the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities.

Available only if all of the following conditions are true:

- **ISDN-BRI Trunks** or **ISDN-PRI** are enabled for the system.
- The trunk group type is access, co, fx, tandem, tie, or wats.
- The **Comm Type** is data, avd, or rbavd.

Valid Entry	Usage
0	Voice and voice-grade data
1	56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
4	64 kbps data on unrestricted channels

Related Links

Comm Type on page 863
Group Type on page 482
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Busy Threshold

Valid Entry	Usage
1 to 255	The number of trunks that must be busy to alert attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code are connected to the attendant. Calls handled by AAR and ARS route patterns go out normally.

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this trunk group generate call detail records. If **Record Outgoing Calls Only** is disabled for CDR, incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • **Abandoned calls**: The system creates a record with a condition code of “H” indicating the time until the call was abandoned. • **Answered calls**: The system creates a record with a condition code of “G” indicating the interval from start of ring to answer. • **Calls to busy stations**: The system creates a record with a condition code of “ I ” indicating a recorded interval of 0.

Related Links

Record Outgoing Calls Only on page 106

CESID I Digits Sent

For emergency 911 service, Communication Manager might send Caller’s Emergency Service Identification (CESID) information to the local telephone company central office or E911 tandem server or switch. This digit string is part of the E911 signaling protocol. Determine the correct entry for this field by talking to the E911 provider. Accepts from one to three digits.

Available only for a cama trunk group.

Related Links

Group Type on page 482

Comm Type

Indicates whether the trunk group carries voice, data, or both.

Note:

Comm Types of avd, rbavd, and data require trunk member ports on a DS1 circuit pack.

Valid Entry	Usage
avd	For applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving end server discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating ends insert a modem pool for any calls. The **Signaling Mode** for the DS1 circuit pack must be set to either common-chan or CAS signaling.
data	All calls across the trunk group originate and terminate at Communication Manager digital data endpoints. Public networks don’t support data: supported by Avaya’s DCP

Table continues…

Valid Entry	Usage
	protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The **Signaling Mode** for the DS1 circuit pack must be set to either robbed-bit or common-chan.
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The **Signaling Mode** for the DS1 circuit pack must be set to robbed-bit unless mixed mode signaling is allowed on the DS1 circuit pack. In that case, the **Signaling Mode** must be isdn-ext or isdn-pri.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use voice.

Related Links

Signaling Mode on page 223

COR

Valid Entry	Usage
0 to 995	The Class of Restriction (COR) for the trunk group. Classes of restriction control access to trunk groups, including trunk-to-trunk transfers. Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed.

Tip:

Remember that FRLs are assigned to classes of restriction. Even if two trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

CO Type

Available only if the country code is 14. Used only by trunk group members administered on a TN464D vintage 2 or later DS1 circuit pack.

Valid Entry	Usage
analog digital	Specifies whether the trunk group is connected to analog or digital facilities at the local telephone company central office (CO).

Related Links

Country on page 864

Country

The country code that corresponds to the protocol used by the local telephone company central office (CO) where the trunk group terminates.

Available only for trunk groups that connect Communication Manager to a CO in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

Valid Entry	Usage
1 to 25, except for 19	For a list of country codes, see the *Country code table*.
11	Communication Manager can be administered for Public Network Call Priority (Call Retention and Re-ring).
15	Communication Manager can be administered for Public Network Call Priority (Intrusion and Re-ring).
18	Communication Manager can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Re-ring).
23	If the trunk Group Type is either CO or DID, Communication Manager can be administered for Block Collect Calls.

Related Links

Country options table on page 783
Trunk Gain on page 893
Trunk Termination on page 894

Dial Access

Controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

Security alert:

Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and are not restricted by facility restriction levels. For security, leave this field disabled unless dial access is needed to test the trunk group.

Valid Entry	Usage
y	Users can gain access to the trunk group by dialing its access code.
n	Users cannot gain access to the trunk group by dialing its access code. Attendants can still select this trunk group with a **Trunk Group Select** button.

Digit Absorption List

Valid Entry	Usage
0 to 4 blank	Assigns a digit absorption list, when used, to a trunk group that terminates at a local telephone company central office.

Note:

In a DCS network, DCS features that use the **remote-tgs** button (on telephones at a remote end) do not work when the incoming trunk group at your end deletes or inserts digits on incoming calls. To manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Direction

The direction of the traffic on this trunk group.

Available for all trunk groups except DID and CPE.

Valid Entry	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing
two-way	Traffic on this trunk group is incoming and outgoing. Required for **Network Call Redirection**.

Related Links

Disconnect Supervision-Out on page 493
Answer Supervision Timeout on page 491
Receive Answer Supervision on page 617

Group Name

A unique name that provides information about the trunk group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the trunk group rather than the group type (DID, WATS).

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Display Character Set on page 787

Group Number

The trunk group number.

Group Type

The type of trunk group. The fields that are displayed and available might change according to the trunk group type selected.

Valid Entry	Usage
Access	Used to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and, therefore, allow

Table continues…

Valid Entry	Usage
	satellite callers unrestricted access to out-dial trunks on the main server. This entry allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks. Used in private networks. This entry allows Inband ANI.
CAMA	Used to route emergency calls to the local community's Enhanced 911 systems.
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
CPE	Used to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Used to direct callers directly to individuals within an organization without going through an attendant or some other central point. This entry allows Inband ANI.
DIOD	Two-way trunks that are used to transmit dialed digits in both directions. In North America, tie trunks are used for applications that require two-way transmission of dialed digits. This entry allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. This entry also allows Inband ANI.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.
ISDN	Used when digital trunks are needed that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Also used for **Network Call Transfer**. **Note:** Available only if **ISDN-PRI**, **ISDN-BRI Trunks**, or both have been enabled for the system. For descriptions of the screens and fields that are unique to ISDN trunks, see *ISDN Trunk Group*.
RLT	Used with Centralized Attendant Service in a private network.
SIP	Used to connect a server running Communication Manager to a Session Manager (SM), or to connect two Communication Manager servers. **Note:** The Automatic CallBack, Priority Calling, and Whisper Page features work incorrectly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Used to connect tandem nodes in a private network. This entry allows Inband ANI.
Tie	Used to connect a server running Communication Manager to a local telephone company central office or to another server or switch in a private network. Tie trunks

Table continues…

Valid Entry	Usage
	transmit dialed digits with both outgoing and incoming calls. This entry also allows Inband ANI.
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can be sent to certain areas (“WATS band”) for a flat monthly charge. Incoming WATS trunks allow toll-free calling to customers and employees.

Related Links

ISDN-PRI on page 801
ISDN-BRI Trunks on page 801
SBS on page 503
Local Country Code on page 306
International Access Code on page 307
Supplementary Service Protocol on page 495
Carrier Medium on page 479
Path Replacement on page 510
Call Still Held on page 880
Call Still Held on page 612

Incoming Budget

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Incoming Budget is the number of incoming calls that a trunk group can allow based on the total call limit.

Communication Manager counts the number of active incoming calls and compares the total number of incoming calls with the incoming call budget. If the total number equals or exceeds the budget, Communication Manager replaces established low-precedence calls with new high-precedence calls and blocks new low-precedence calls.

Entire trunk budget (Number of Members) = Incoming budget + Outgoing budget

Valid Entry	Usage
0–255	The number of incoming calls that the trunk group can receive. The total of the **Incoming Budget** and the **Outgoing Budget** fields must not be greater than the total number of trunk group members.
Blank	This field is blank by default.

Incoming call budget high threshold

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap when the system reaches the incoming call budget high threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Incoming call budget low threshold

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap to clear a budget threshold alarm when the system reaches the incoming call budget low threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Incoming Destination

Sets the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination entered here is also the default night service destination. Available only for incoming or two-way trunk groups.

Valid Entry	Usage
Valid extension number	Calls go to this extension number. You can enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it is active.

Table continues…

Valid Entry	Usage
	Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.
blank	Leave this field blank if **Trunk Type (in/out)** is not administered as auto/….

Related Links

Direction on page 481
Night Service on page 870
Trunk Type on page 894

ITC

Determines the line coding the Generalized Route Selection feature uses for comparison to select appropriate routes for voice and data calls.

Available only when the **Comm Type** is data, avd, or rbavd and a **BCC** value is administered.

Valid Entry	Usage
rest(tricted)	Restricted trunks use ami-basic or ami-zcs line coding and can carry only restricted calls.
unre(stricted	Unrestricted trunks use b8zs, hdb3, or cmi line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC can have only unrestricted trunks as members.

Related Links

Comm Type on page 863

Night Service

Sets the destination for incoming calls when **Night Service** is operating. If a night service destination is administered for an individual trunk that is a member of this group, that entry overrides the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:

Whenever possible, use a night service destination on your switch; otherwise, some features do not function correctly over a DCS network.

Valid Entry	Usage
An extension number (can be a VDN)	The extension of your night service destination.

Table continues…

Valid Entry	Usage
	Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.
blank	Leave this field blank if **Trunk Type (in/out)** is not administered as auto/….

Related Links

Trunk Type (in/out) on page 876

Number of Members

The system displays this field only for SIP trunk groups.

Valid entry	Usage
1 to 255	The number of SIP trunks that are members of the Trunk group.

Related Links

Group Type on page 482

Outgoing Budget

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Outgoing Budget is the number of outgoing calls a trunk group can allow within the total limit.

Communication Manager counts the number of active outgoing calls, and compares the total number of outgoing calls with the outgoing call budget. If the total number equals or exceeds the budget, Communication Manager replaces established low-precedence calls with new high-precedence calls and blocks new low-precedence calls.

Valid Entry	Usage
0–255	The number of outgoing calls that the trunk group can make. The total of the **Incoming Budget** and the **Outgoing Budget** fields must not be greater than the total number of trunk group members.
Blank	The field is blank by default.

Note:

If the total number of **Incoming Budget** and **Outgoing Budget** is less than **Number of Members**, the system displays the following message:

```
WARNING: Total budget allocated less than total number of trunk group
members.
```

Outgoing call budget high threshold

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap when the system reaches the outgoing call budget high threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Outgoing call budget low threshold

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap to clear a budget threshold alarm when the system reaches the outgoing call budget low threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Outgoing Display

With Outgoing Display, telephones can show the name and number of the trunk group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Prefix-1

If enabled, the prefix “1” is added to the beginning of the digit string for outgoing calls. Use this field for outgoing and two-way trunk groups handling long distance service. Do not enable for trunk groups in AAR or ARS route patterns.

Available only for CO, FX, and DIOD trunk groups.

Protocol Type

The type of line signaling protocol used by the local telephone company central office for DID and DIOD trunk groups.

Available only if the **Country** code is 15. Used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack.

Valid Entry	Usage
inloc	Incoming local. Only the inloc protocol provides ANI.
intol	Incoming toll.

Related Links

Country on page 864
Country options table on page 783

Queue Length

Available only for outgoing or two-way trunk groups.

Valid Entry	Usage
1 to 100	The number of outgoing calls that can wait in queue when all trunks in a trunk group are busy. Calls wait in queue in the order in which they were made. If a queue is administered, a caller hears a confirmation tone when no trunk is available for the outgoing call. The caller can then disconnect and wait; when a trunk becomes available, Communication Manager calls the extension that placed the original call. Communication Manager remembers the number the caller dialed and automatically completes the call.
0	Callers receive a busy signal when no trunks are available. Use for DCS trunks.

Related Links

Direction on page 481

Service Type

The service for which this trunk group is dedicated. In addition to the predefined services or features listed as valid entries, any previously administered user-defined Network Facility **Facility Type** of 0 (feature) or 1 (service) is allowed.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — A subscriber can receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, **Dial Access** can be enabled.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.

Table continues…

Comments on this document? infodev@avaya.com

Valid Entry	Usage
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related Links

Facility Type on page 600

Signaling Group

Available only for sip trunk groups.

Valid Entry	Usage
1 to 999	The number of the SIP signaling group associated with this trunk group.

Related Links

Group Type on page 482
Group Number on page 675

TAC

The trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group. The characters "*" and "# " can be used as the first character in a TAC. Accepts a one- to four-digit number.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Toll Restricted

If enabled, restricts toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.

Trunk Flash

Enables or disables multifunction telephones to access local telephone company central office (CO) customized services that are provided by servers at the far-end or CO. These CO customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the Communication Manager telephone on an active trunk call.

Trunk Signaling Type

Controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field also controls the signaling used by members in public network digital trunk groups.

Available only for access, aplt, rlt, tandem, or tie trunk groups.

E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling must match the signaling type administered on the far-end server. Use these values only when all trunk group members are assigned to ports on a TN464F, TN2464, or TN2140 circuit pack.

Valid Entry	Usage
cont	Continuous
dis	Discontinuous

The following entries are for tie trunks in Main-Satellite/Tributary networks. Use these values only when all trunk group members are assigned to a TN497 circuit pack.

Valid Entry	Usage
tgu	For outgoing trunks, tgu at the main server running Communication Manager administers a tie trunk group connected to a satellite server. (This same group should be administered as tge at the satellite.)
tge	For incoming trunks, tge at a satellite server administers a tie trunk group connected to the main server running Communication Manager. (This same group should be administered as tgu at the main server.)
tgi	For internal trunks, tgi administers a two-way tie trunk group between two satellites or between the main server and a satellite. (This trunk group should be administered as tgi on both servers.)

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. The following entries are for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.

Valid Entry	Usage
cont	Continuous E&M signaling.
pulsed	Pulsed E&M signaling.
discont	Discontinuous E&M signaling.
blank	Leave blank for R2 signaling.

Related Links

Group Type on page 482

Trunk Type (in/out)

Controls the seizure and start-dial signaling used on this trunk group. Settings might differ for incoming and outgoing trunks.

Valid Entry	Usage
auto	Used for immediate connection to a single preset destination (incoming local telephone company central office trunks, for example). No digits are sent, because all calls terminate at the same place.
cont	Continuous signaling is used with Italian E&M tie trunks. The server/switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer.
delay	The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it is ready to receive the digits.
disc	Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The Avaya server can seize a trunk by sending a single, short signal for the duration specified by the **Normal Outgoing Seize Send** value. However, with the Three-Way Seizure option, the calling end can also send routing information to the called end by sending one or a series of brief seizure signals.
wink	The sending server or switch does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.
immed	The sending server or switch sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac 2-wire-dc 3-wire	Used with local telephone company central office (CO) trunks in Russia. Select the type of connection to the CO. Check with the network service provider for the type of connection. To use these entries, the **Country** code must be 15 and the CO trunks must use ports on a TN2199 circuit board.

Tip:

When incoming trunks use the setting immed/immed, the far-end server seizes the trunk and sends digits without waiting for acknowledgment from the receiving end. When traffic is heavy, the receiving server or switch might not immediately attach a Touch Tone Receiver to a call and, therefore, lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end server or switch to avoid this problem.

Related Links

Country on page 864
Normal Outgoing Seize Send (msec) on page 917

Version

Adjusts the signaling on multi-country local telephone company central office (CO) trunk circuit packs. Entries adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country.

Available only for CO, FX, and WATS trunk groups when the **Country** code is 5, 16, or 23.

If the **Country** code is 5, this field controls only TN2147 ports.

Valid Entry	Usage
a	Uses standard signaling for the Netherlands public network.
b	Uses country 1 (U.S.) signaling. This value is appropriate if Communication Manager is connected to a CO using an Ericcson AXE-10 switch.

If the **Country** code is 16 or 23, this field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.

Valid Entry	Usage
a	Sets input impedance to 600 Ohms.
b	Sets input impedance to 900 Ohms. This value is appropriate in Brazil.

Related Links

Country on page 864

Trunk Group: page 2

```
change trunk-group 1                                        Page   2 of  22
      Group Type: co                    Trunk Type:

TRUNK PARAMETERS

    Outgoing Dial Type: tone                                Cut-Through? n
     Trunk Termination: rc                  Disconnect Timing(msec): 500

            Auto Guard? n     Call Still Held? n      Sig Bit Inversion: none
     Analog Loss Group: 6                            Digital Loss Group: 11
                                Trunk Gain: high

 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 10             Receive Answer Supervision? n
           Administer Timers? y
              XOIP Treatment: auto
```

Administer Timers

Enables or disables administration of timers on this trunk group. The default for the ISDN trunk group type is disabled. All other trunk group types are enabled by default.

Available for all trunk group types *except* cpe, h.323, and sip.

Related Links

Group Type on page 482

Analog Loss Group

Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid Entry	Usage
1 to 17	The index into the loss plan and tone plan. If values are administered other than in between 6 and 10 or 15 and 17, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

Note:

This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers.

Related Links

Receive Answer Supervision on page 617
Administer Timers on page 490

Auto Guard

Enables or disables Auto Guard, which prevents repeated seizures of a defective trunk. Communication Manager does a maintenance busy-out on these trunks. This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports use a trunk defective if no dial tone is detected on an outgoing call, and the **Outpulse Without Tone** feature is disabled for the system.

Available only for co or fx trunk groups.

Related Links

Group Type on page 482
Outpulse Without Tone on page 310

Bit Rate

Available when the **Comm Type** is avd or rbavd. Also available if the **Comm Type** is data, but only if **ISDN-PRI** is enabled for the system.

Valid Entry	Usage
300 1200 2400 4600 9600 19200	The baud rate to be used by pooled modems.

Related Links

Comm Type on page 863
ISDN-PRI on page 801

Caller ID for Service Link Call to H.323 1xC

This field is available only when you set the **Direction** field to `outgoing` or `two-way` and the trunk group type is `ISDN` or `SIP`.

Valid entry	Usage
original-calling-number	Before sending the call to the service link, Communication Manager changes the calling party number to the original calling party number. When the user receives an incoming call, the number that the mobile displays is the original caller ID and not the extension number.
station-extension	Communication Manager does not change the calling party number. The default value is station-extension.

Call Still Held

If enabled, the system prevents glare by extending the Incoming Glare Guard timer and delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call. This field is used when the receiving end media server or switch initiates the disconnection of incoming calls. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the **Country** code is 2.

Available only for co or fx trunk groups.

Related Links

Group Type on page 482

Cut-Through

Available only if the **Outgoing Dial Type** field is rotary or tone.

Security alert:

Enabling this field reduces the ability to prevent toll fraud.

Valid Entry	Usage
y	Users get dial tone directly from the far end of the trunk. If you set the **Cut-Through** field to y, some administered restrictions are bypassed.
n	Users receive switch dial tone. Instead of digits being sent to the central office, they are collected, and sent all at once when the user finishes dialing.

Related Links

Outgoing Dial Type on page 889

Cyclical Hunt

Controls the starting point Communication Manager uses to search for an available trunk when a call is sent to the trunk group. Cyclical hunts can be enabled or disabled at any time, however, all the trunks in the group must be idle or busied out.

Available only for two-way loop-start trunks.

Valid Entry	Usage
y	Communication Manager starts its search from the last trunk seized. This method is faster, and therefore suited for high-traffic trunk groups.
n	Communication Manager starts each search at member 1 (the first trunk administered as a group member).

Related Links

Direction on page 481
Trunk Type on page 894

Delay Call Setup When Accessed Via IGAR

The system displays this, when the **Group Type** field is isdn or sip. If a user in one network region accesses an ISDN or SIP trunk in another network region through IGAR, the **Delay Call Setup When Accessed Via IGAR** field determines whether the IGAR trunk and the outgoing trunk are initiated in sequence or parallel.

Valid Entry	Usage
y	The outgoing trunk call waits until the IGAR trunk call is active.
n	The two trunk calls are set up in parallel. The default value is n.

Dial Detection

Indicates whether digit pulses are detected by observing the A-wire (default) or the B-wire only. Applies only to TN2199 ports. The **Country** code must be 15.

Related Links

Country on page 864

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group. If values other than 18 or between 11 and 15 are administered, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Digits

If the **Digit Treatment** is absorption, this field specifies how many digits are deleted. If the **Digit Treatment** is insertion, this field identifies the specific digits that are added.

Valid Entry	Usage
1 to 5	The number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	The actual digits to be added (inserted).
blank	This field can be blank only if **Digit Treatment** is not administered.

Related Links

Digit Treatment on page 882

Digit Treatment

Modifies an incoming digit string by adding or deleting digits. This is required if the number of digits received does not match the dial plan.

Requires administration of the **Digits** to add or delete.

Valid Entry	Usage
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.
blank	The incoming digit string is not changed.

Related Links

Digits on page 882

Disconnect Supervision-In

Indicates whether Communication Manager receives disconnect supervision for incoming calls over this trunk group.

Available only for incoming or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end server or switch sends a release signal when the calling party releases an incoming call, and

Table continues…

Valid Entry	Usage
	the far-end server or switch is responsible for releasing the trunk. Enhances Network Call Redirection.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on incoming calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

In general, U.S. local telephone company central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with the network services provider.

Related Links

Trunk Direction on page 617
Direction on page 481

Disconnect Supervision-Out

Indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. Available for outgoing or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end sends a release signal when the called party releases an outgoing call, and the far-end is responsible for releasing the trunk. Enhances Network Call Redirection. Available only if **Answer Supervision Timeout** is 0 and **Receive Answer Supervision** is enabled.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on outgoing calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

Verify that the far-end server or switch provides answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with the network services provider.

Related Links

Direction on page 481
Answer Supervision Timeout on page 491
Receive Answer Supervision on page 617

Disconnect Timing (msec)

Valid Entry	Usage
140 to 2550 ms in increments of 10	Specifies the minimum time in milliseconds that the local telephone company central office or far-end server requires to recognize that this server has disconnected from a call. This timer does not affect ports on a circuit pack that uses the administrable

Valid Entry	Usage
	Incoming Disconnect and Outgoing Disconnect timers. Settings on those two timers override this field. The default of 500 is an industry standard and should not be changed. If this field is set too high, the server or switch does not disconnect sometimes when it should. If this field is set too low, the server or switch disconnects when it should not.

Disconnect Type

Indicates which side or user controls the disconnect. A refers to the calling party, and B refers to the called party.

Available only if the **Country** code is 15 and the **Trunk Type** is 2-wire-ac, 2-wire-dc, or 3-wire. Applies only to the TN2199 port.

Valid Entry	Usage
AandB	Both parties control the disconnect.
AorB	Either party controls the disconnect.

Related Links

Country on page 864
Trunk Type on page 894

Drop Treatment

Valid Entry	Usage
intercept busy silence	Determines what the calling party hears when the called party terminates an incoming call. For security reasons, you must apply a tone; silence can provide an opening for hackers. Applies only to DID trunks.

Note:

In Italy, **Drop Treatment** must be administered as intercept for all DID trunk groups.

Duplex

Available if the **Comm Type** is avd or rbavd. Also available if the **Comm Type** is data, but only if **ISDN-PRI** is enabled for the system.

Note:

Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid Entry	Usage
full	Allows simultaneous two-way transmission, which is most efficient.
half	Supports only one transmission direction at a time.

Related Links

Comm Type on page 863
ISDN-PRI on page 801

End-to-End Signaling

Available only for a cpe (customer-provided equipment) trunk group.

Valid Entry	Usage
60 to 360 ms in increments of 10	The duration of the touch tone signal sent to auxiliary equipment. Equipment such as paging equipment and music sources might be connected to Communication Manager by auxiliary trunks. Communication Manager might send DTMF signals (touch tones) to these devices. This field sets the duration of these tones.

Related Links

Group Type on page 482

Expected Digits

Valid Entry	Usage
1 to 18	The number of digits that the far-end server sends for an incoming connection. If the end is absorbing digits on this trunk group, the entry in this field must be larger than the value administered for the number of digits to be absorbed.
blank	Required if **Digit Treatment** is insert and **Digits** is a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table. If left blank, digit absorption cannot be administered.

Related Links

Digits on page 882
Digit Treatment on page 882

Extended Loop Range

Available only for a DID trunk group. This field is used only with the TN459A circuit pack.

Valid Entry	Usage
y	The distance between the local telephone company central office (CO) and the server is greater than the required distance.
n	The distance between the CO and the server is not greater than the required distance.

Format

Specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number.

Available only if the **Send Calling Number** or **Send Connected Number** field is enabled or restricted.

Valid Entry	Usage
public	Indicates that the number plan according to CCITT Recommendation E. 164 is used.
unknown	Indicates the **Numbering Plan Indicator** is unknown.
private	Indicates the **Numbering Plan Indicator** is PNP.
unk-pvt	Determines the type of number from the private numbering format, but the **Numbering Plan Indicator** is unknown.

Related Links

Send Calling Number on page 504
Send Connected Number on page 505

Group Type

Displays the type of trunk group.

Related Links

Group Type on page 482
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Incoming Calling Number - Delete

Available only for incoming or two-way trunk groups.

Valid Entry	Usage
1 to 15 all blank	The number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.

Related Links

Direction on page 481

Incoming Calling Number - Format

The TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message excludes CPN information and digits are added, the outgoing message contains these digits. If a numbering format is not administered in this case, the value defaults to pub-unk. If the numbering format is administered as unknown, the trunk group is modified to unk-unk encoding of the TON/NPI. Therefore, this field also must contain a value other than unknown.

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid Entry	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Related Links

Format on page 885
Format on page 499

Incoming Calling Number - Insert

Valid Entry	Usage
0 to 9 all blank	Up to 15 digits added to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Incoming Dial Tone

Indicates whether or not the server running Communication Manager gives dial tone in response to far-end seizures of the trunk group.

Valid Entry	Usage
y	Used if the incoming trunk group transmits digits. For example, this option is used for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Used for trunks that are not sending digits, such as tandem or incoming local telephone company central office trunks.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. This value should match what the local telephone company central office provides.

Available for Access, APLT, DID, DIOD, DMI-BOS, FX, RLT, Tandem, or WATS trunk groups. Also available for tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Use this entry for Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. With this entry, trunk group supports both DTMF and rotary signals. Use this entry for the Inband ANI feature. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
rotary	Allows only the dial pulse addressing method used by non-touch tone telephones. Though the tone entry supports rotary dialing as well, it is inefficient to reserve touch tone registers for calls that do not use DTMF.
mf	Use this option if the **Trunk Signaling Type** field is blank. The system displays this field only if you set the **Multifrequency Signaling** field on the System Parameters Customer-Options screen to y. You cannot use this option if the **Used for DCS** field is y. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
mf2/6	Use this entry for signaling in Spain. This signaling type provides normal public network service and a virtual private network digital trunk service called IBERCOM. Use this entry if the **Trunk Signaling Type** field is blank. The system displays this entry only if you enable **Multifrequency Signaling** for the system. The system does not display this entry if the trunk is used for DCS.

Related Links

Group Type on page 482

Incoming Rotary Timeout (sec)

Valid Entry	Usage
5 to 99 blank	Sets the maximum time to wait to receive all incoming digits from the far-end switch.

Related Links

Incoming Dial Type on page 887

Line Length

Available only for trunk groups with a **Trunk Signaling Type** of tge, tgi, or tgu.

Valid Entry	Usage
short long	The line length.

Related Links

Group Type on page 482
Trunk Signaling Type on page 875

Outgoing Dial Type

Sets the method used to transmit digits for an outgoing call. Usually, this method should match what the local telephone company central office provides.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Available for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. Also available for Tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Uses Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. Allows the trunk group to support both DTMF and rotary signals.
rotary	Allows only the dial pulse addressing method used by non-touch tone telephones. For example, this value is appropriate for an internal full touch tone system and for a connection to a local telephone company central office that only supports rotary dialing.
automatic	For tie trunks if the **Trunk Signaling Type** is not administered. This provides “cut-through” operation to outgoing callers who dial a trunk access code, connecting them directly to local telephone company central office dial tone and bypassing any toll restrictions administered on Communication Manager.
mf	Used if a **Trunk Signaling Type** is not administered. Available only if **Multifrequency Signaling** is enabled on the **System-Parameters Customer-Options** screen. Not available if this trunk is used for DCS. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
mf2/6	Used if a **Trunk Signaling Type** is not administered. Available only if **Multifrequency Signaling** is enabled on the **System-Parameters Customer-Options** screen. Not available if this trunk is used for DCS.
r1mf	Enter r1mf for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter r1mf to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. **Group type** field must be set to co.

Related Links

Group Type on page 482
Multifrequency Signaling on page 802
Trunk Signaling Type on page 875

Preferred Minimum Session Refresh Interval (sec)

The system displays this field only for the SIP trunk groups that do not support Seamless Converged Communications Across Network (SCCAN) calls.

Use this field to set the value of the session refresh timer of non-SCCAN applications. The timer starts after Communication Manager establishes a session with the far-end SIP entities. After every

timer interval, Communication Manager sends a session refresh request as a Re-INVITE or an UPDATE message to the SIP entities. If the SIP entities do not receive the request before the interval period expires, the session ends.

Valid entry	Usage
90 to 64800	Defines the session refresh timer. The default value is 600.

Note:

If the SIP trunk is a member of a trunk group connected to Session Manager 6.3.6 or earlier, ensure that the value of the field is not more than 1800. If the SIP trunk is not a member, configure Session Manager with an adaptation module for Communication Manager, and set the **SessionTimeout** parameter on System Manager to a value greater than the value in the **Preferred Minimum Session Refresh Interval** field.

Related Links

Group Type on page 482
SCCAN on page 891

Receive Answer Supervision

If enabled, the Communication Manager provides answer supervision for a trunk group. For Outbound Call Management applications, use for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Related Links

Answer Supervision Timeout on page 491
Administer Timers on page 490

Receive Release Ack

If enabled, the **Receive Release Ack** field is used to set the receive forward and backward release ack options on TN2140 ports (used for Italian and Hungarian tie trunks). Available only if the **Trunk Signaling Type** is cont or dis.

Related Links

Trunk Signaling Type on page 875

Redirect on OPTIM failure

Valid Entry	Usage
250 to 32000 milliseconds	Calls are redirected by ROOF (Redirect on OPTIM Failure) procedures if a call terminates to an Extension to Cellular user cell phone (of any application type) that does not have an associated office telephone (or other method of terminating to a physical telephone), or the office telephone is out of order and Communication Manager receives a disconnect prior to the call being answered. The procedure redirects the call so that the caller does not hear ringback forever.

SCCAN

If enabled, this trunk group provides support for incoming SCCAN calls. Available only for sip type trunk groups when **Enhanced EC500** is enabled for the system.

Related Links

Enhanced EC500 on page 799
Group Type on page 482

Send Answer Supervision

If enabled, Communication Manager signals the calling server when an incoming call is answered. Available only if the **Trunk Signaling Type** is cont or dis. The field applies only to TN2140 ports. Available only for incoming or two-way trunks.

Related Links

Direction on page 481
Trunk Signaling Type on page 875

Send Release Ack

Indicates whether a release acknowledgment is sent in response to a forward or backward release signal. Available only if the **Trunk Signaling Type** is cont or dis. This field applies only to TN2140 ports (used for Italian and Hungarian tie trunks).

Related Links

Trunk Signaling Type on page 875

Sig Bit Inversion

Indicates which bits in bit-oriented signaling should be inverted, if any. For trunk ports on TN2242 and TN464B and later circuit packs, this field inverts the A- and B-bits as necessary so that the far-end server or switch can understand seizure and release signals from Communication Manager. If the far-end server, such as a local telephone company central office, on this trunk group interprets the A- and B-bits differently from the default, invert one or both bits — to change “1” to “0” and vice-versa in the A-bit.

Valid Entry	Usage
A B A&B none	For the TN464B and later circuit packs.
A and none	For the Japanese 2Mbit trunk circuit pack.

Related Links

Country Protocol on page 211

Supplementary Service Protocol

Available only for ISDN trunk groups.

Valid Entry	Usage
a	Allows ASAI Flexible Billing. AT&T, Telcordia Technologies, Nortel. When the **Country** code for the DS1 circuit pack is 1A, SSA selects AT&T custom supplementary services. When the **Country** code for the DS1 circuit pack is 1B, SSA selects Telcordia Technologies Supplementary Services. When the **Country** code for the DS1 circuit pack is 1C, SSA selects Nortel Proprietary Supplementary Services.
b	QSIG; also used for SBS signaling trunk groups when full QSIG functionality is needed.
c	ETSI; used for Network Call Deflection.
d	ECMA QSIG
e	Allows ASAI Flexible Billing. Allows DCS with rerouting when the trunk is used for DCS and **DCS with Rerouting** is enabled.
f	Feature Plus
g	ANSI. Available only if **ISDN-PRI** or **ISDN-BRI Trunks** is enabled for the system, or the trunk is used for DCS. Used for Network Call Transfer.

Related Links

Country Protocol on page 211
DCS with Rerouting on page 797
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801
Used for DCS on page 911

Synchronization

Available only if the **Group Type** is :

- dmi-bos or isdn
- access, co, fx, tandem, tie, or wats; and the **Comm Type** is avd or rbavd
- access, co, fx, tandem, tie, or wats; the **Comm Type** is data; and **ISDN-PRI** or **ISDN-BRI Trunks** is enabled for the system

⚠ Caution:

Do not change this field without the assistance of Avaya or your network service provider.

Valid Entry	Usage
async	Determines whether the trunk group uses synchronous or asynchronous communications.

Valid Entry	Usage
sync	

Related Links

Comm Type on page 863
Group Type on page 482
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801

Trunk Gain

Specifies the amplification applied to the trunks in this group. With the values administered for **Trunk Termination** and **Country** code, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO and DID circuit packs are set automatically to high.

Valid Entry	Usage
high	Used if users complain of low volume.
low	Used if users complain of squeal or feedback.

Related Links

Country on page 864
Trunk Termination on page 894

Trunk Hunt

Defines the trunk hunt search order. Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Valid Entry	Usage
ascend	Enables a linear trunk hunt search from the lowest to highest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.
cyclical	Enables a circular trunk hunt based on the sequence the trunks were administered within the trunk group. When using ISDN-BRI interfaces, only cyclical is supported. The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations.
descend	Enables a linear trunk hunt search from the highest to lowest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.

Related Links

Wideband Support on page 508

Trunk Termination

Adjusts the impedance of the trunk group for optimal transmission quality.

Valid Entry	Usage
600ohm	The distance to the local telephone company central office (CO) or to the server at the other end of the trunk is less than 3,000 feet.
rc	The distance to the CO or to the server at the other end of the trunk is more than 3,000 feet.

Trunk Type

Controls the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and must match the corresponding setting on the far-end server or switch.

Available only for CO, DID, FX, and WATS trunk groups.

Valid Entry	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible. Ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, loop-start signaling is used only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto auto/delay auto/immed auto/wink	The term before the slash tells Communication Manager how and when it receives incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming central office trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it is ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch. • wink — The sending server running Communication Manager does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.
2-wire-ac 2-wire-dc 3-wire	These entries are used with local telephone company central office (CO) trunks in Russia. The specific CO should match one of these values. Available only if the **Country** code is 15 and the CO trunks use ports on a TN2199 circuit board.

Related Links

Country on page 864

Comments on this document? infodev@avaya.com

Unicode Name

Used to determine whether to send Name1 (legacy name) or Name2 (Unicode name). The value for this field is only examined for calls to SIP stations Available only for sip trunk groups.

Note:

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Valid Entry	Usage
y	Uses the Unicode name.
n	Uses the name as specified on the station form. **Note:** Any non-ASCII characters in the name do not appear correctly on a SIP phone.
auto	The choice to use the station name or the Unicode name is automatically determined based on the called phone's capability and the user's preference.

Related Links

Group Type on page 482

Wink Timer (msec)

Sets the wink timer as follows:

- Sets the maximum duration of the wink signal (wait-for-wink-to-end) when **Trunk Type (in/out)** is administered as .../wink.
- Sets the maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start) when **Trunk Type (in/out)** is administered as .../delay.

Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds. While some circuit packs are hard-coded to allow the full 5 seconds in both cases, other circuit packs allow you reduce the allowed start time and duration, therefore, reducing the window in which glare can occur.

Available only for wink-type trunks.

Related Links

Trunk Type (in/out) on page 876

XOIP Treatment

Valid Entry	Usage
auto modem fax	Identifies treatment of an incoming trunk call. When set to modem, fax, or tty, the serving digital signal processor algorithms is placed in immediate pass-through mode. Use this field when the incoming trunk is transporting the XOIP traffic. By identifying this external equipment through administration, VoIP firmware uses this field to

Valid Entry	Usage
tty	determine if it is necessary to immediately attempt to put a call in pass-through mode so that the DSPs do not miss identification of the initial tone events. Default is auto. **Note:** • This feature does not apply to IP trunks (SIP and H.323 trunks). • This field is intended for exception cases only. For the majority of trunk groups, use the default setting of auto.

Trunk Group: page 3

Caution:

Do not change fields on this page without assistance from Avaya or your network service provider.

```
change trunk-group 1                                    Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n              Measured: none
                                                    Maintenance Tests? y
                                Data Restriction? n

   Abandoned Call Search? n
   Suppress # Outpulsing? n

       Charge Conversion: 1
           Decimal Point: none
       Currency Symbol:
           Charge Type: units       Receive Analog Incoming Call ID: disabled
                                          Per Call CPN Blocking Code:
                                        Per Call CPN Unblocking Code:

          Outgoing ANI:                        Ds1 Echo Cancellation? n
```

Abandoned Call Search

Indicates whether this trunk group conducts an Abandoned Call Search to identify ghost calls. Abandoned Call Search is designed to work with analog ground-start local telephone company central office (CO) trunks that do not provide disconnect supervision. The CO must support Abandoned Call Search for the feature to work properly. If the CO provides disconnect supervision, the Abandoned Call Search feature is not needed.

Available only for ground-start type trunks.

Related Links

Trunk Type on page 894

ACA Assignment

Indicates whether Automatic Circuit Assurance (ACA) measurements are taken for this trunk group.

Charge Conversion

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related Links

Direction on page 481
Trunk Direction on page 617

Charge Type

Text string used to describe charges related to a telephone call. These words or characters appear on telephone displays after the charge amount. Typically uses either the currency symbol or the charge type, but not both. Accepts up to seven characters. Embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups.

Related Links

Direction on page 481

Connected to CO

Enables or disables overlap sending to a local telephone company central office (CO).

Available only for tie trunk groups.

Related Links

Group Type on page 482

Currency Symbol

The telephone displays the symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for ISDN trunk groups, if the Charge Advice field is set to a value other than none.

Related Links

Direction on page 481
Trunk Direction on page 617

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Decimal Point

The appropriate representation for a decimal point as telephone displays it. Available only with outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field is independent whether the calling display shows a decimal point. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related Links

Direction on page 481
Charge Advice on page 480

DS1 Echo Cancellation

Enables or disables echo cancellation on a per port basis. If enabled, reduces voice call echo.

Note:

Changes to the DS1 Echo Cancellation field is incomplete until one of the following occurs:

- Port is busied-out or released.
- Trunk group is busied-out or released.
- SAT command test trunk group is performed.
- Periodic maintenance runs.

DSN Term

Enables or disables the trunk group as a DSN termination telephone. The default is disabled.

Glare Handling

Determines the reaction of Communication Manager to glare.

The following circuit packs can detect glare:

- TN767 (all releases)
- TN760C (or later releases)
- TN464C (or later releases)

Available only for two-way trunks when the outgoing side of the **Trunk Type** is either …/`wink` or …/`delay`.

Valid Entry	Usage
control	Communication Manager seizes the trunk and proceeds with call setup. The other switch finds another trunk.
backoff	The other server or switch seizes the trunk and proceeds with call setup. The server or switch finds another trunk.
none	Not administered.

Related Links

Direction on page 481
Trunk Type on page 894

Hold/Unhold Notifications

If enabled, hold and unhold messages are sent over the isdn trunk when a user places a call on hold/unhold. Default is enabled.

Available only for isdn trunk groups.

Related Links

Group Type on page 482

Incoming Tone (DTMF) ANI

Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the Intuity Audix and ASAI interfaces, and displayed on the telephone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Available only if the **Incoming Dial Type** is tone.

Valid Entry	Usage
*ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The telephone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The telephone displays Call from 555-3800.
no	Not administered.

Related Links

Incoming Dial Type on page 887

Internal Alert

Indicates if internal ringing and coverage is used for incoming calls.

Invoke ID for USNI Calling Name

Communication Manager uses the **Invoke ID for USNI Calling Name** field to select an Invoke ID each time a calling name is sent over the ISDN trunk group using the USNI protocol.

The system displays the **Invoke ID for USNI Calling Name** field only when:

- The **Group Type** field is set to `ISDN`.
- The **Carrier Medium** field is set to `PRI/BRI`.
- The **Supplementary Service Protocol** field is set to `a`.

However, the valid entries, `variable` and `fixed-1`, have an impact only when the following conditions are met:

- On the Feature-related system parameters screen, the **USNI Calling Name for Outgoing Calls** field is set to `y`.
- On the Trunk group screen, the **Send Name** field is set to `y`.

Valid entry	Usage
variable	Communication Manager selects a new Invoke ID each time a calling name is sent over the ISDN trunk group. This entry is the default value.
fixed-1	Communication Manager selects an Invoke ID with the fixed value 1 each time a calling name is sent over the ISDN trunk group.

Long Holding Time (hours)

Available only if Automatic Circuit Assurance measurements are taken for the trunk group.

Valid Entry	Usage
0 to 10	The number of hours that the system considers a long holding time. A value of `0` indicates that the system does not support long holding calls.

Related Links

ACA Assignment on page 497

Maintenance Tests

Enables or disables hourly maintenance tests on this trunk group.

Available only for aplt, isdn, sip, or tie trunk groups.

Related Links

Group Type on page 482

Measured

Indicates if the system transmits data for this trunk group to the Call Management System.

Valid Entry	Usage
internal	Sends the data to the Basic Call Management System (BCMS), the VuStats data display, or both. Available only if **BCMS (Basic)** or **VuStats** is enabled for the system.
external	Sends the data to the CMS.
both	Collects data internally and sends it to the CMS. Available only if **BCMS (Basic)** or **VuStats** is enabled for the system.
none	Trunk group measurement reports are not required.

Related Links

VuStats on page 812
BCMS (Basic) on page 807

MF Tariff Free

If enabled, Communication Manager generates an MFC Tariff-Free Backward Signal during call setup instead of the “free” signal. This aids local telephone company central office billing.

Available only for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the **Incoming Dial Type** is mf or for tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis, and the **Incoming Dial Type** is mf.

Related Links

Group Type on page 482
Incoming Dial Type on page 887
Trunk Signaling Type on page 875

Modify Tandem Calling Number

The system displays this field for the SIP and ISDN trunk groups. This field is available with outgoing or two-way trunks.

Valid Entry	Usage
natl-intl-prefix	Adds the national or international prefixes from the Locations Parameters screen when the calling party number is appropriate.
tandem-cpn-form	Modifies the calling party number IE in the previously administered format specified for the Tandem Calling Party Number.
no	Does not modify the calling party number.

Related Links

Modify Tandem Calling Number on page 501

Network Call Redirection

Valid Entry	Usage
deflect	Allows Network Call Deflection.
ANSI-transfer	Allows Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Allows Network Call Transfer for MCI DMS 250 switches.
telcordia-tbct	Allows Network Call Transfer for Lucent 5ESS or Nortel DMS100 switches.

Numbering Format

Available only for sip trunk groups.

Valid Entry	Usage
public	Indicates that the number plan according to CCITT Recommendation E. 164 is used.
private	Indicates the **Numbering Plan Indicator** is PNP.

Outgoing ANI

The digit string sent in place of normal ANI. Overrides the normal ANI if this trunk group is used for an outgoing call with ANI. The ANI is sent exactly as administered, except for the normal truncation to seven digits for Russian ANI. This ANI override works both for calls originated in Communication Manager and calls tandemed through it. Accepts up to 15 digits.

Available only for CO, DIOD, FX, and WATS trunk groups.

Path Replacement Method

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system. Unavailable if **Path Replacement with Retention** is enabled.

Valid Entry	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.

Table continues…

Valid Entry	Usage
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if **Path Replacement with Retention** is enabled. Path replacement fails (and the original trunk group is retained) if **Path Replacement with Retention** is disabled.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 495
Supplementary Services with Rerouting on page 814
DCS with Rerouting on page 797
Path Replacement with Retention on page 510

Path Replacement with Retention

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system.

Valid Entry	Usage
y	Retains the original trunk group.
n	Provides path replacement according to settings for the **Path Replacement Method** .

Related Links

Supplementary Service Protocol on page 495
Supplementary Services with Rerouting on page 814
DCS with Rerouting on page 797
Group Type on page 482

PBX ID

Valid Entry	Usage
1 to 63 blank	Identifies the remote switch in the network with which the trunk communicates on a DCS signaling link. Available only for trunks that are used for DCS.

Related Links

Used for DCS on page 911

Per Call CPN Blocking Code

A one- to four-digit number used for access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only. The characters * and # can be used as the first digit.

Per Call CPN Unblocking Code

A one- to four-digit number used for access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only. The characters * and # can be used as the first digit.

Precedence Incoming

Available only when the trunk group is a DSN termination telephone and the trunk group type is tie.

Valid Entry	Usage
digit	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as digits (rotary pulses).
dtmf (a-d)	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as DTMF signals (touchtones).

Related Links

DSN Term on page 499
Group Type on page 482

Precedence Outgoing

Available only when the trunk group is a DSN termination telephone and the trunk group type is tie.

Valid Entry	Usage
digit	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as digits (rotary pulses).
dtmf (a-d)	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as DTMF signals (touchtones).

R2 MFC Signaling

Available only if:

- **Multinational Locations** is enabled for the system, and either of the following:
  - **Outgoing Dial Type** is mf.
  - **Incoming Dial Type** or **Outgoing Dial Type** is rotary, and **Country** code is 15 (Russia).

Valid Entry	Usage
1 to 8	The MFC signaling parameters set used by this trunk group.

Related Links

Country on page 864
Multinational Locations on page 803
Incoming Dial Type on page 887
Outgoing Dial Type on page 889

Receive Analog Incoming Call ID

Enables or disables the collection of incoming call ID information on analog trunks. Fifteen characters of name and number information associated with an incoming call (ICLID, or incoming call line identification information) is stored and displays.

Available for CO, DID, and DIOD trunk groups when **Analog Trunk Incoming Call ID** is enabled. The trunk must be incoming or two-way.

Valid Entry	Usage
Bellcore	Collects ICLID information in the U.S.
NTT	Collects ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.
V23–Bell	For Telcordia Technologies protocol with V.23 modem tones. Used in Bahrain and similar countries.

Related Links

Analog Trunk Incoming Call ID on page 794
Direction on page 481

Replace Unavailable Numbers

If enabled, replaces unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. Applies to BRI/PRI, H.323, and SIP trunks. Also applies to analog trunks if **Analog Trunk Incoming Call ID** is enabled for the system and **Receive Analog Incoming Call ID** for the trunk is set to any value except disabled.

Available only if the group type is isdn or sip.

Related Links

Analog Trunk Incoming Call ID on page 794
Group Type on page 482
Receive Analog Incoming Call ID on page 904

Request Category

Indicates if Communication Manager should request a call category from the local telephone company central office (CO).

Available only if the **Country** code is 15 and MF shuttle signaling is enabled.

Related Links

Country on page 864
Shuttle on page 909

Seize When Maintenance Busy

Indicates whether this server generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end server or switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end server or switch from reporting problems with a trunk that has been removed from service. This field does not affect the behavior of the far-end server or switch. It controls the behavior of your server and defines the expected far-end behavior.

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. For DIOD trunks using TN464F (or later release) or TN2464 circuit packs, available only for diod trunk groups when the **Trunk Signaling Type** is pulsed, cont, or dis.

Valid Entry	Usage
near-end	Communication Manager generates an outgoing seizure when a trunk is maintenance busied, but the far-end server or switch does not. The seizure is maintained until the maintenance busyout is released.
far-end	The far-end server or switch generates an outgoing seizure when a trunk is maintenance busied, but this server running Communication Manager does not.
both-ends	Both this server running Communication Manager and the far-end server or switch generate an outgoing seizure when a trunk is maintenance busied.

If a server generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end server or switch, perhaps leading to a far-end maintenance busy out, unless the far-end server or switch is administered to expect this behavior.

If the administered value of this field is either far-end or both-ends, any abnormally long incoming seizure (including failure to drop from a completed call) is the result of a far-end maintenance busy condition. This assumption might be incorrect, since the abnormally long seizure might actually be due to failure of the trunk circuit.

Related Links

Group Type on page 482
Trunk Signaling Type on page 875

Send Called/Busy/Connected Number

Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is enabled for the trunk group.

Valid Entry	Usage
y	The dialed number is sent on incoming or tandemed ISDN calls. This field must be enabled in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the dialed number on incoming or tandemed ISDN calls.
r	Restricted. The connected number is sent “presentation restricted”.

Related Links

Numbering — Public/Unknown Format on page 604
QSIG Value-Added on page 511
Numbering-Private Format on page 602

Send Calling Number

Specifies whether the calling party’s number is sent on outgoing or tandemed ISDN calls.

Note:

The Numbering - Public/Unknown Format can override the Send Calling Number administration

Valid Entry	Usage
y	The calling party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the calling party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The calling number is sent “presentation restricted”. If set to restricted, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Related Links

Numbering-Private Format on page 602
Incoming Number Format on page 96
Numbering — Public/Unknown Format on page 604

Send Connected Number

Specifies if the connected party’s number is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is disabled for the trunk group.

Valid Entry	Usage
y	The connected party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used. This field must be enabled for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.
n	Disables the sending of the connected party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The connected number is sent “presentation restricted”. If this field is set to r, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination for the DS1 Circuit Pack, you should not administer the Send Connected Number as restricted, as

this causes display problems. The Numbering - Public/Unknown Format overrides the Send Connected Number administration for any administrable block of extensions.

Related Links

Numbering-Private Format on page 602
QSIG Value-Added on page 511
Numbering — Public/Unknown Format on page 604

Send EMU Visitor CPN

Controls which calling party identification (extension of the primary telephone or extension of the visited telephone) is used when a call is made from a visited telephone. There are areas where public network trunks disallow a call if the calling party information is invalid. In this case, there can be instances where the extension of the primary telephone is invalid and the extension of the visited telephone must be used.

Valid Entry	Usage
y	Sends calling party identification information on the extension of the EMU user's telephone.
n	Sends calling party identification information on the primary telephone.

Send Name

Specifies whether the calling, connected, called, or busy party's administered name is sent to the network on outgoing or incoming calls. Available only for isdn or sip trunk groups.

Valid Entry	Usage
y	When the **Supplementary Service Protocol** is e (DCS with Rerouting), only values of y and n are permitted.
n	When the **Supplementary Service Protocol** is e (DCS with Rerouting), only values of y and n are permitted.
r (restricted)	The calling/connected name will be sent by Communication Manager, but will be marked "presentation restricted". This value is valid only if the **Supplementary Service Protocol** is a (national supplementary service), b (for called/busy only), or d for the QSIG Global Networking Supplementary Service Protocol.

Note:

If name information is not administered for the calling station or the connected, called, or busy station; the system sends the extension number instead of the name.

Related Links

Group Type on page 482
Supplementary Service Protocol on page 892

Short Holding Threshold

Available only if Automatic Circuit Assurance measurements are taken for this trunk group.

Valid Entry	Usage
0 to 30	The number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk. If 0 is entered, no short holding calls are recorded.

Related Links

ACA Assignment on page 497

Short Holding Time (seconds)

Available only if Automatic Circuit Assurance measurements are taken for this trunk group.

Valid Entry	Usage
0 to 160	The length of time that the system considers as being a short holding time. If 0 is entered, the system does not support short holding calls.

Related Links

ACA Assignment on page 497

Show ANSWERED BY on Display

Available only for isdn pri/bri, isdn h.323, and sip trunk groups.

Valid Entry	Usage
y	The words “ANSWERED BY” display in addition to the connected telephone number on calls over this trunk. This is the default. **Note:** Based on display language settings for stations, “ANSWERED BY” is translated into and displayed in the appropriate language.
n	Only the connected telephone number displays. This might be preferred when outgoing calls are over a trunk that might be redirected.

Shuttle

Enables or disables MF shuttle signaling. It can be administered on TN464D (or later release) or TN2199 circuit packs.

Available only for co, fx, or wats trunk groups, with a **Country** code of 15, when the **Outgoing Dial Type** is rotary.

Related Links

Country on page 864
Group Type on page 482
Outgoing Dial Type on page 889

Signaling Group

The signaling group number.

Start B Signal

Indicates which B-signal should be used to start a call. The value administered in this field must be coordinated with the local telephone company central office.

Available only when the **Country** code is 15 and MF shuttle signaling is enabled.

Valid Entry	Usage
1	Start calls with signal B1 (first digit).
2	Start calls with signal B2 (next digit).
3	Start calls with signal B3 (previous digit).

Related Links

Country on page 864
Shuttle on page 909

Start Position

Available only when the **Country** code is 15 and MF shuttle signaling is enabled.

Valid Entry	Usage
1 to 9	Indicates which digit in the digit string is the previously sent digit. The value administered in this field must be coordinated with the local telephone company central office.

Related Links

Country on page 864
Shuttle on page 909

Suppress # Outpulsing

Indicates whether or not to suppress the final “#” in cases where the system would normally outpulse it. Used if end-to-end signaling begins with (and includes) “#”. This field should be enabled when the local telephone company central office or any other facility treats “#” as an error.

Time (sec) to Drop Call on No Answer

Available only for co or diod trunk groups when the **Outgoing Dial Type** is mf, or for co, diod, fx, or wats trunk groups when the **Country** code is 15.

Valid Entry	Usage
0 to 1200	The duration in seconds that Communication Manager should wait for outgoing calls to be answered. If the call is unanswered in the specified number of seconds, the call drops.

Valid Entry	Usage
	If 0 is entered, the timer is not set and no calls drop.

Related Links

Country on page 864
Group Type on page 482
Outgoing Dial Type on page 889

Used for DCS

If enabled, this trunk group sends and receive messages on a DCS signaling link.

Note:

This field cannot be enabled if the trunk group number is greater than 255 or if the Trunk Access code is more than 3 digits long.

Related Links

TSC Supplementary Service Protocol on page 688
DCS with Rerouting on page 797
ISDN-BRI Trunks on page 801
ISDN-PRI on page 801
Group Type on page 675

Used Only for Paging

Indicates whether or not this trunk is used only for paging. Default is disabled.

Available only for wats trunk groups when **Port Network Support** is disabled for the system.

Related Links

Group Type on page 482
Port Network Support on page 804

Voice Paging Timeout (sec)

Available only if the trunk is used only for paging.

Valid Entry	Usage
10 to 6000	The number of seconds before a paged trunk call drops. Default is 10.

Related Links

Used Only for Paging on page 911

Wideband Support

Note:

This feature is not supported on the DS1 interfaces on branch gateways (G700/G350).

Enables or disables wideband switching on this trunk group. Only trunk members from TN464C or later circuit packs can use wideband switching.

Available only if **Wideband Switching** is enabled for the system.

Note:

Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Related Links

Wideband Switching on page 807

Administrable Timers

The system might not display this screen for all trunk group types.

Caution:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

```
change trunk-group 1                                          Page   4 of  22
 ADMINISTRABLE TIMERS
               Send Incoming/Outgoing Disconnect Timers to TN465 Ports? n

                                          Outgoing Dial Guard(msec): 1600
  Incoming Glare Guard(msec): 1500      Outgoing Glare Guard(msec): 1500


      Ringing Monitor(msec): 5200            Incoming Seizure(msec): 500
   Outgoing End of Dial(sec): 10       Outgoing Seizure Response(sec): 5
 Programmed Dial Pause(msec): 1500
          Flash Length(msec): 540




END TO END SIGNALING
    Tone(msec): 350      Pause(msec): 150

OUTPULSING INFORMATION
     PPS: 10     Make(msec): 40    Break(msec): 60      PPM? n
```

Answer Send (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is dis. Only TN2140 and TN2199 ports receive this timer.

Valid Entry	Usage
10 to 2550 in increments of 10	The duration of the answer signal pulse.

Related Links

Direction on page 481
Trunk Type on page 894

Busy Tone Disconnect

If enabled, Communication Manager recognizes a busy tone signal as a disconnect on this trunk group.

Available only if **Enable Busy Tone Disconnect for Analog loop-start Trunks** is enabled for the system country options.

Related Links

Enable Busy Tone Disconnect for Analog Loop-start Trunks on page 787

Cama Outgoing Dial Guard (msec)

Available only for cama trunk groups (the trunk group type used for emergency 911 service).

Valid Entry	Usage
25 to 6375 in increments of 25	Minimum interval between the seizure acknowledgment on the receiving server or switch and the outpulsing of digits by this server.

Related Links

Group Type on page 482

Cama Wink Start Time (msec)

Available only for cama trunk groups.

Valid Entry	Usage
20 to 5100 in increments of 20	The duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Related Links

Group Type on page 482

Disconnect Signal Error (sec)

The system displays this field only for ground-start trunk groups.

Valid entry	Usage
1 to 255 in increments of 1	The maximum interval that Communication Manager waits to receive a disconnect signal from the far-end after the local party, such as a telephone or tie trunk, goes on-hook. If the timer expires, the disconnect signal fails.

Flash Length (msec)

Valid Entry	Usage
10 to 2550 in increments of 10	The duration of a flash signal generated toward the local telephone company central office. This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), TN753 (if **Country** code is 23), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Glare

Available only for two-way or outgoing trunk groups when the **Trunk Type** is cont. Only TN2140 ports receive this timer.

Valid Entry	Usage
40 to 100 in increments of 10	The minimum acceptable interval between the moment your server running Communication Manager sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Related Links

Direction on page 481
Trunk Type on page 894

Incoming Dial Guard (msec)

Valid Entry	Usage
10 to 2550 in increments of 10	The minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. Communication Manager does not accept digits before this timer expires. This timer is never sent to TN429 ports.

Incoming Disconnect (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is blank or cont.

Valid Entry	Usage
50 to 2550 in increments of 10	The minimum valid duration of a disconnect signal for an incoming call. Communication Manager does not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use 600.

Related Links

Direction on page 481
Trunk Type on page 894

Incoming Disconnect Send (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
500 to 1200 in increments of 100	The duration of the backward release signal the server running Communication Manager sends at the end of an incoming call.

Related Links

Direction on page 481
Trunk Type on page 894

Incoming Glare Guard (msec)

Available only for two-way trunk groups.

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This delay gives the far-end time to release all equipment after the trunk is released. This field cannot be blank.

Related Links

Direction on page 481

Incoming Incomplete Dial Alarm (sec)

Valid Entry	Usage
1 to 255 in increments of 1	The maximum acceptable interval between an incoming seizure and receipt of all digits. Intervals greater than this limit generate an inline error. Only the TN436 (all), TN459 (all), TN464C (or later), TN767, TN2140, TN2146, TN2184, TN2199, and TN2242 circuit packs use this timer.

Incoming Partial Dial (sec)

Available only if the **Incoming Dial Type** is rotary. This timer is never sent to TN429 ports.

Valid Entry	Usage
5 to 255 in increments of 1	The maximum time allowed between incoming rotary digits.

Related Links

Incoming Dial Type on page 887

Incoming Seizure (msec)

Available only for incoming or two-way trunk groups, and, when applicable, the **Trunk Type** is cont. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the shortest incoming seizure signal the server running Communication Manager can recognize. For ICLID, set this field to 120. The field cannot be blank.

Related Links

Direction on page 481
Trunk Type on page 894

Normal Outgoing Seize Send (msec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
10 to 990 in increments of 10	The duration of the signal the server running Communication Manager sends for an outgoing seizure.

Related Links

Direction on page 481
Trunk Type on page 894

Outgoing Dial Guard (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after lifting the receiver, but before the timer expires, is not outpulsed until the timer expires. This field cannot be blank.

Outgoing Disconnect (msec)

Valid Entry	Usage
50 to 2550 in increments of 10	The minimum valid duration of a disconnect signal for an outgoing call. Communication Manager does not recognize shorter disconnect signals. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. This field cannot be blank. For Brazil pulsed E&M signaling, use 600.

Outgoing Disconnect Send (msec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
100 to 9900 in increments of 100	The duration of the forward release signal the server running Communication Manager sends at the end of outgoing calls.

Related Links

Direction on page 481
Trunk Type on page 894

Outgoing End of Dial (sec)

Available for outgoing or two-way trunk groups when the network does not provide answer supervision.

Valid Entry	Usage
1 to 254 in increments of 1	The maximum time Communication Manager waits to receive answer supervision for outgoing calls on the ports controlled by firmware timers. Controls firmware answer supervision timers on circuit packs that have them. During a cut-through operation, timing begins after Communication Manager sends each outgoing digit and ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. For Brazil pulsed E&M signaling, use 40.

Related Links

Direction on page 481
Receive Answer Supervision on page 617

Outgoing Glare Guard (msec)

Available only for outgoing and two-way trunk groups.

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This delay gives the far-end time to release all equipment after the outgoing trunk is released. This field cannot be blank.

Related Links

Direction on page 481

Outgoing Last Digit (sec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis or cont. Only TN497 and TN2140 ports receive this timer.

Valid Entry	Usage
1 to 40 in increments of 1	The maximum time that Communication Manager waits for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Related Links

Direction on page 481
Trunk Type on page 894

Outgoing Rotary Dial Interdigit (msec)

Available only if:

- The trunk **Group Type** is access, aplt, co, diod, dmi-bos, fx, rlt, tandem, or wats, and the **Outgoing Dial Type** is rotary.
- The trunk **Group Type** is tie, the **Trunk Type** is blank, cont, or dis, and the **Outgoing Dial Type** is rotary.
- The trunk **Group Type** is tie, and the **Trunk Type** is tge, tgi, or tru.

Valid Entry	Usage
150 to 2550 in increments of 10	The minimum time between outpulsed digits on outgoing rotary trunks.

Related Links

Group Type on page 482
Outgoing Dial Type on page 889
Trunk Type on page 894

Outgoing Seizure (msec)

Available only if the **Country** code is 15, for outgoing or two-way trunk groups, when the **Trunk Type** is 2-wire-ac, 2-wire-dc, or 3-wire. This timer is sent only to the TN2199 circuit pack.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the outgoing seizure signal.

Related Links

Country on page 864
Direction on page 481
Trunk Type on page 894

Outgoing Seizure Response (sec)

Valid Entry	Usage
1 to 255 in increments of 1	The maximum interval that Communication Manager waits after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received within this time, a seizure failure response is uplinked. This timer is sent to the TN438B, TN439, TN447, TN458, TN464B (or later), TN465B (or later), TN767, TN2140, TN2147, TN2184, TN2199, and TN2242 circuit packs. For Brazil pulsed E&M signaling, use 255.

Programmed Dial Pause (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The exact duration of the pause used during abbreviated dialing, ARS outpulsing, and terminal dialing operations. This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All central office (CO) circuit packs that accept administrable timers accept this timer.

Release Ack Send (msec)

Available only for incoming and two-way trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
500 to 1200 in increments of 100	The duration of the signal the server running Communication Manager sends for a forward release acknowledgment. After the server running Communication Manager receives a forward release signal, it must send a forward release acknowledgment signal.

Related Links

Direction on page 481
Trunk Type on page 894

Ringing Monitor (msec)

The minimum time that Communication Manager requires to determine if a trunk disconnects.

You cannot leave the field blank.

Valid entry	Usage
200 to 51000 in increments of 200	If the ringing signal disappears for a duration longer than the time specified in this field, you can be sure that the call is disconnected. This timer is sent to TN464C (or later), TN767, TN438 (all), TN447, TN465 (all), TN2138, TN2147, TN2184, and TN2199 CO circuit packs.

Seize Ack Delay (msec)

Available only for incoming or two-way trunk groups if the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
40 to 120 in increments of 10	The maximum interval the server running Communication Manager waits after receipt of an incoming seizure to send seizure acknowledgment.

Related Links

Direction on page 481
Trunk Type on page 894

Seize Ack Send (msec)

Available only for incoming or two-way trunk groups if the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
10 to 990 in increments of 10	The duration of the seizure acknowledgment signal the server running Communication Manager sends in response to an incoming seizure.

Related Links

Direction on page 481
Trunk Type on page 894

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

If enabled, the incoming disconnect and outgoing disconnect timer values are sent to the trunk group ports that are on a TN465 board. Available only for a co, fx, or wats trunk group.

END TO END SIGNALING

Pause (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The interval (pause) between DTMF tones sent from a hybrid telephone. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Related Links

Trunk Type on page 894

Tone (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the DTMF tone sent when a button on a hybrid telephone is pressed. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Related Links

Trunk Type on page 894

OUTPULSING INFORMATION

Break (msec)

The duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.

Valid Entry	Usage
20 to 80 in increments of 5	If the **PPS** (pulses per second) value is 10, the sum of the **Make (msec)** and **Break (msec)** values must equal 100.
10 to 40 in increments of 5	If the **PPS** (pulses per second) value is 20, the sum of the **Make (msec)** and **Break (msec)** values must equal 50.

Related Links

Make (msec) on page 923
PPS on page 924

Frequency

Identifies the Periodical Pulse Metering (PPM) pulse frequency, or frequencies, sent by the public network. Circuit packs can detect up to three different frequencies (12kHz, 16kHz, and 50Hz), plus two frequency combinations (50Hz/12kHz and 50Hz/16kHz). This field controls TN465B, TN2138, and TN2184 circuit packs. Available for outgoing or two-way trunk groups if **PPM** is enabled.

Valid Entry	Usage
12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if 12k is administered, the circuit pack will be set to detect 12kHz.
16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if 16k is administered, the circuit pack will be set to detect 16kHz.
50	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if 50 is administered, the circuit pack will be set to detect 16kHz.
50/12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if 50/12k is administered, the circuit pack will be set to detect 12kHz.
50/16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if 50/16k is administered, the circuit pack will be set to detect 16kHz.

Related Links

Direction on page 481
PPM on page 923

Make (msec)

The duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.

Valid Entry	Usage
20 to 80 in increments of 5	If the **PPS** (pulses per second) value is 10, the sum of the **Make (msec)** and **Break (msec)** values must equal 100.
10 to 40 in increments of 5	If the **PPS** (pulses per second) value is 20, the sum of the **Make (msec)** and **Break (msec)** values must equal 50.

Related Links

Break (msec) on page 922
PPS on page 924

PPM

Determines if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. For CO, DIOD, FX, PCOL, and WATS trunks. Available only for outgoing or two-way trunk groups.

Related Links

Frequency on page 923
Direction on page 481

Direction on page 481

PPS

Available only for cama trunk groups.

Valid Entry	Usage
10 20	The rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group. The TN439, TN458, TN497, TN747Bv12 (or later), and TN767 circuit packs send rotary pulses at 10 pps only.

Related Links

Break (msec) on page 922
Make (msec) on page 923

Trunk Group: ATMS Thresholds

Available for outgoing or two-way trunk groups when **ATMS** is enabled for the system.

Caution:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

```
change trunk-group 1                                          Page   5 of  22
                                ATMS THRESHOLDS
    TTL Type: 105-w-rl                  Far End Test No:
  TTL Vendor:                             TTL Contact:
Trunk Vendor:                           Trunk Contact:
Trunk Length:

                                MARGINAL        UNACCEPTABLE
                                Min   Max         Min   Max
                  1004 Hz Loss: -2    21          -2    21

                                -Dev  +Dev        -Dev  +Dev
                   404 Hz Loss: 9     9           9     9
                  2804 Hz Loss: 9     9           9     9

          Maximum C Message Noise:    55              55
          Maximum C Notched Noise:    74              74
                    Minimum SRL-HI:   0               0
                    Minimum SRL-LO:   0               0
                       Minimum ERL:   0               0

              Allow ATMS Busyout, Error Logging and Alarming? n
```

Related Links

ATMS on page 796
Direction on page 481

Far-End Test No

The access number dialed to reach the terminating test line (TTL). Accepts up to 16 digits.

Trunk Contact

The name or telephone number of someone from the trunk vendor who can be contacted if there are problems with the trunks. Accepts up to 25 alphanumeric characters.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information might be useful.

Valid Entry	Usage
0 to 4 k	The length of the trunk group in kilometers.
0 to 4 m	The length of the trunk group in miles.

Trunk Vendor

The name of the vendor providing service over this trunk group that needs to be notified in the event of problems with the trunks in this trunk group. Accepts up to 22 alphanumeric characters.

TTL Contact

The name or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line. Accepts up to 25 alphanumeric characters.

TTL Type

Type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and which threshold values need to be administered.

Valid Entry	Usage
105-w-rl	105 with return loss. Full range of 18 measurements or some defaults for return loss used (56A).
105-wo-rl	105 without return loss. Cannot return default values for far-end return loss.
high-lts	High-level tone source. Sends a fixed sequence of tones at 0 dBm.
low-lts	Low-level tone source. Sends a fixed sequence of tones at -16dBm.
100	100 type. Up to 5 measurements that sends a 1004 Hz tone, then a quiet termination.
102	102 type. One measurement that sends a 1004 Hz tone.

TTL Vendor

The name of the vendor supplying the terminating test line (TTL). Accepts up to 22 alphanumeric characters.

MARGINAL / UNACCEPTABLE

Allow ATMS Busyout, Error Logging and Alarming

Enables or disables ATMS error logging and alarming.

Marginal Threshold --Dev - 404 Hz Loss

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold -+Dev - 404 Hz Loss

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold --Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold -+Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid Entry	Usage
0 to 21	The maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid Entry	Usage
15 to 55	The maximum C-message noise telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid Entry	Usage
34 to 74	The maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Min -1004 Hz Loss

Valid Entry	Usage
–2 to 21	The minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Minimum ERL

Valid Entry	Usage
0 to 40	The minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid Entry	Usage
0 to 40	The minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid Entry	Usage
0 to 40	The minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Available only if **Allow ATMS Busyout**, **Error Logging**, and **Alarming** are enabled.

Valid Entry	Usage
0, 25, 50, 75, 100	The highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Related Links

Allow ATMS Busyout, Error Logging and Alarming on page 926

Unacceptable Threshold --Dev - 404 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold -+Dev - 404 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold --Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold -+Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid Entry	Usage
0 to 21	The maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid Entry	Usage
15 to 55	The maximum C-message noise interference in dBmC above reference noise terminating on a telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid Entry	Usage
34 to 74	The maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid Entry	Usage
–2 to 21	The minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Minimum ERL

Valid Entry	Usage
0 to 40	The minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

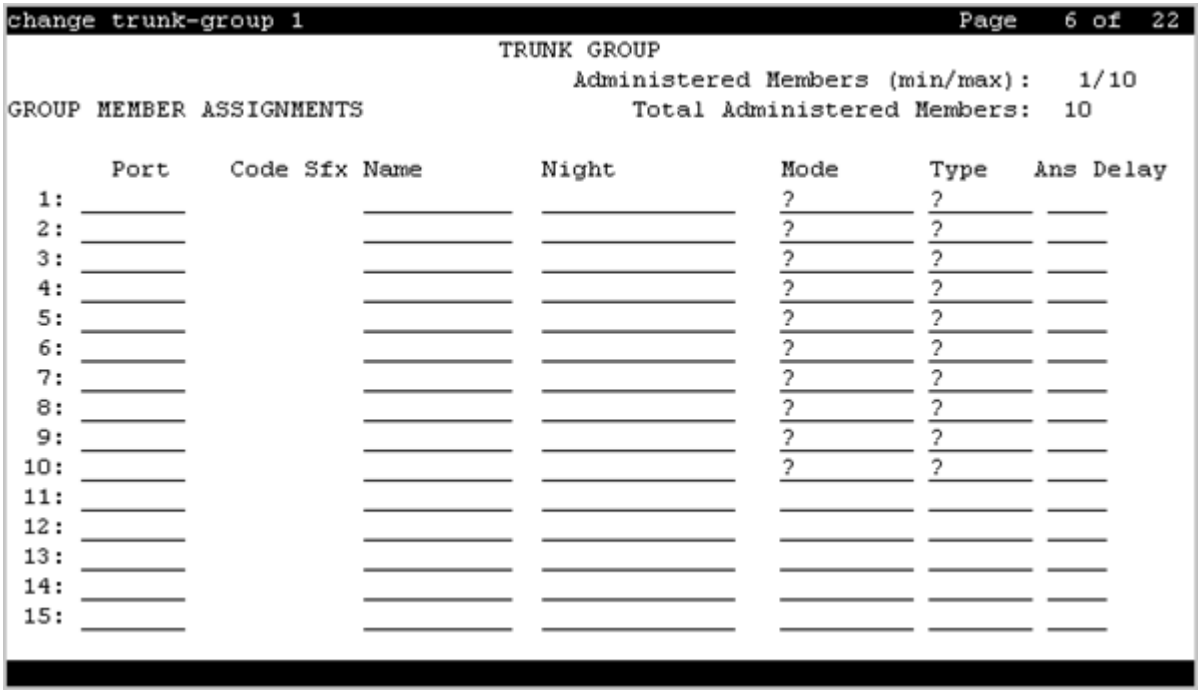
Valid Entry	Usage
0 to 40	The minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid Entry	Usage
0 to 40	The minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Trunk Group: Group Member Assignments

For SIP trunks, the group member-assignment pages are not individually administered. The system automatically populates and displays these fields based on the number of members of SIP trunk groups previously specified.



```
change trunk-group 1                                          Page   6 of  22
                                  TRUNK GROUP
                                       Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:  10

      Port     Code Sfx Name         Night             Mode      Type    Ans Delay
 1:  _______           _____________ _________________ ?         ?       ____
 2:  _______           _____________ _________________ ?         ?       ____
 3:  _______           _____________ _________________ ?         ?       ____
 4:  _______           _____________ _________________ ?         ?       ____
 5:  _______           _____________ _________________ ?         ?       ____
 6:  _______           _____________ _________________ ?         ?       ____
 7:  _______           _____________ _________________ ?         ?       ____
 8:  _______           _____________ _________________ ?         ?       ____
 9:  _______           _____________ _________________ ?         ?       ____
10:  _______           _____________ _________________ ?         ?       ____
11:  _______           _____________ _________________ _________ _______ ____
12:  _______           _____________ _________________ _________ _______ ____
13:  _______           _____________ _________________ _________ _______ ____
14:  _______           _____________ _________________ _________ _______ ____
15:  _______           _____________ _________________ _________ _______ ____
```

Administered Members (min/max)

The minimum and maximum member numbers that have been administered for this trunk group.

Ans Delay

Caution:

Customers should not attempt to administer this field. Go to the Avaya Support website at http://support.avaya.com for current documentation, product notices, knowledge articles on administering this field.

Valid Entry	Usage
20 to 5100 in increments of 20	The length of time the server running Communication Manager waits before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.
blank	Same as setting the field to zero.

Code

The type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address, the field is blank.

Mode

The signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip-switch settings on the circuit pack.

Caution:

Customers should not attempt to administer this field. Go to the Avaya Support website at http://support.avaya.com for current documentation, product notices, knowledge articles on administering this field.

Valid Entry	Usage
e&m	For six-wire connections that pair two signaling wires with four voice wires. This configuration is used in the vast majority of systems in the United States.
simplex	For four-wire connections that do not use an additional signaling pair. This configuration is very rare in the United States.

Name

The name of the trunk group member. The name should identify the trunk unambiguously. Accepts up to 10 characters.

Example

- The telephone number assigned to incoming trunks
- The Trunk Circuit Identification number assigned by the service provider

Night

The night service destination for this trunk group member if different from the night service destination administered for the trunk group. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid Entry	Usage
A valid extension	The extension of the night destination for the trunk.
attd	Calls go to the attendant when night service is active.
blank	Not administered

Related Links

Night Service on page 870

Port

Valid Entry	Usage
1 to 64	First and second characters are the cabinet number.
A to E	Third character is the carrier.
0 to 20	Fourth and fifth characters are the slot number.
01 to 04 (Analog TIE trunks) 01 to 31	Six and seventh characters are the circuit number.
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Sfx

The model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address, the field is blank.

Total Administered Members

The total number of members administered in the trunk group.

Type

Caution:

Customers should not attempt to administer this field. Go to the Avaya Support website at http://support.avaya.com for current documentation, product notices, knowledge articles on administering this field.

Valid Entry	Usage
t1-stan t1-comp t5-rev type-5	Specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the **Trunk Type** is cont), TN437, TN439, TN464 with any suffix, or TN458 circuit packs. Available only if the **Trunk Type** is blank or cont. t5-rev is allowed only for the TN760D vintage 10 or later. When **Type** is t5 rev, **Mode** must be e&m.

Related Links

Trunk Type on page 894

Trunk Group: Protocol Variations

Available only for sip trunk groups.

```
change trunk-group 1                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                          Mark Users as Phone? n
                 Prepend '+' to Calling Number? n
           Send Transferring Party Information? n
                      Network Call Redirection? n
                         Send Diversion Header? y
                       Support Request History? n
                  Telephone Event Payload Type:


             Convert 180 to 183 for Early Media? n
      Always Use re-INVITE for Display Updates? n
            Identity for Calling Party Display: P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                                   Enable Q-SIP? n
```

Related Links

Group Type on page 482

Accept Redirect to Blank User Destination

The following table represents the way in which this field works.

Valid entry	User destination in the Refer-To domain of the SIP request	Result
y	present	Communication Manager places the call to the user destination.
y	missing	Communication Manager places the call over the signaling group specified in the Proxy Sel Rte Pat field of the Locations screen.
n	present	Communication Manager places the call to the user destination.
n	missing	Communication Manager rejects the SIP request.

Always Use re-INVITE for Display Updates

The system displays this field only when the **Group Type** field is sip.

Valid entry	Usage
y	In SIP messages, Communication Manager sends the re-invite message to display update.
n	In SIP messages, Communication Manager sends the update message to display update. The default value is n.

Block Sending Calling Party Location in INVITE

Based on the value of this field, Communication Manager selects the IP address that must be included in the Via header in an outgoing call. The system displays this field only if the value of the **Group Type** field in the trunk group screen is sip

Valid Entry	Usage
y	Communication Manager does not send the IP address of the caller in the INVITE
n	Communication Manager sends an IP address that identifies the location of the caller. This IP address is included in the lower-most Via header in the INVITE. Based on the type of endpoint of the caller, Communication Manager determines the IP address to be sent. • For calls made from H.323 endpoints, Communication Manager sends the IP address of the endpoint. • For calls made from TDM endpoints or trunks, Communication Manager sends the IP address of the media processor used in the call. This value is the default value.

Build Refer-To URI of REFER From Contact For NCR

The system displays this field only for SIP trunk groups for which **Network Call Redirection** is enabled. Communication Manager uses the REFER message during a call transfer.

Valid entry	Usage
y	Communication Manager uses the address in the Contact header to create the URI of the Refer-To header in a REFER message.
n	Communication Manager uses the address in the From header to create the URI of the Refer-To header in a REFER message.

Convert 180 to 183 for Early Media

Used for early media and direct media cut-through. When SDP answer is returned by Communication Manager in 18x messages, some entities have a problem receiving the SDP in a 180 Ringing message. If you set the **Convert 180 to 183 for Early Media** field to y, Communication Manager puts the SDP into a 183 Session Progress message. The default value is n.

Enable Q-SIP

The system displays Enable Q-SIP, only when the **Group Type** field is sip.

Valid Entry	Usage
y	Enables the QSIG over SIP (Q-SIP) feature for the trunk group. If trunk members are already assigned to this trunk group, you must not change the value of **Enable Q-SIP** field. If you change the value of **Enable Q-SIP** field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
n	The QSIG over SIP feature for the trunk group is disabled. By default, the value is n.

Identity for Calling Party Display

The system displays this field, when the **Group Type** field is sip.

This field determines which header to retrieve display information for the calling party when both the From and P-Asserted-Identity headers are available.

Valid Entry	Usage
P-Asserted-Identity	If a call is terminating at SIP, H.323, or DCP stations, or at outgoing H.323 or SIP trunk, Communication Manager displays the name and Calling Party number from the PAI header instead of the FROM header. This is the default value.

Table continues…

Comments on this document? infodev@avaya.com

Valid Entry	Usage
From	If a call is terminating at SIP, H.323, or DCP stations, or at outgoing H.323 or SIP trunk, Communication Manager displays the name and Calling Party number from the FROM header instead of the PAI header.

Interworking of ISDN Clearing with In-Band Tones

The setting of this field is for call-clearing of the tandem calls in which the incoming trunk to Communication Manager is a SIP trunk and the outgoing trunk from Communication Manager is an ISDN trunk. The far end at the ISDN PRI trunk ends the call.

If you select the **keep-channel-active** option, Communication Manager keeps the call active so that the caller can hear an in-band tone or announcement played over the ISDN trunk.

If you select the **drop-with-sip-error** option, Communication Manager detects the cause of call clearing, maps ISDN Cause Value to a SIP error response message, and sends the response message to the caller. With the **drop-with-sip-error** option, an adjunct placing outgoing calls is informed that the call has ended and the user can disconnect the call immediately instead of waiting for the ISDN network to end the call after the tone or announcement is complete.

Note:

An in-band tone is an announcement or a tone that provides call-clearing information.

Valid entry	Usage
keep-channel-active	Communication Manager sends a SIP 183 Progress message to the caller. The ISDN network plays an in-band tone that the caller hears. The default value is keep-channel-active.
drop-with-sip-error	Communication Manager sends a SIP error response message, which specifies the cause of call-clearing to the caller. The called party plays an in-band tone. The caller does not hear the in-band tone or announcement.

Mark Users as Phone

Enables or disables the encoding of URIs in call control signaling messages originated at the gateway with the “user=phone” parameter. No subscription messages are encoded with the “user=phone” parameter, even when the field is set to y. Default is n.

Note:

Do not change the default of n for this field unless you are sure that every recipient of SIP calls using this trunk can accept and properly interpret the optional “user=phone” parameter. Enterprise users without support for “user=phone” in their SIP endpoints will experience adverse effects, including rejected calls.

Network Call Redirection

If enabled, Network Call Redirection (NCR) service is signaled over this trunk group. NCR only works on trunk groups connected to Service Providers that support NCR.

Prepend '+' to Calling/Alerting/Diverting/Connected Number

The system displays this field only for SIP trunk groups.

If you set this field to y, when a call is routed over a trunk group, Communication Manager inserts a leading plus (+) sign, if absent, in outgoing calling, alerting, diverting, or connected numbers. The default value is n.

To prevent Communication Manager from inserting a leading plus (+) sign in the calling number when a call is routed from Communication Manager to Session Manager, set the **Numbering Format** field to private and the **Prepend '+' to Calling/Alerting/Diverting/Connected Number** field to n.

Related Links

Group Type on page 482

QSIG Reference Trunk Group

The system displays this, only when the **Group Type** field is sip and the **Enable Q-SIP** is set to y.

Valid Entry	Usage
1 to 2000	(For Avaya S8510 Server) Assigns a number for the QSIG trunk group. If trunk members are already assigned to this trunk group, you must not change the value of **QSIG Reference Trunk Group** field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
1 to 99	(For Avaya S8300D or S8300E Server) Assigns a number for the QSIG trunk group. If trunk members are already assigned to this trunk group, you must not change the value of **QSIG Reference Trunk Group** field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
blank	No QSIG trunk group is assigned. By default, the value is blank.

Send Diversion Header

This field is available only when the **Service Type** field is set to public-ntwrk. If you enable the **Send Diversion Header** field, the SIP diversion header is sent on the public SIP trunks. This header

allows service providers to use the ten digit number of the forwarding party for functions such as billing. Customers can administratively enable or disable use of the SIP diversion header on any public SIP trunk. The default value is n.

Related Links

Service Type on page 486

Send Transferring Party Information

Enables or disables sending the transferring party information on a transferred call. Default is disabled.

Support Request History

You can activate the **Support Request History** field for an Off PBX Station (OPS) SIP endpoint to capture the location of the request that was sent to a proxy. This information will be used to send diversion information and to notify of the origin of the call to a calling off pbx station SIP Endpoint. If this field is disabled, History-Info header will not be included in any requests.

Telephone Event Payload Type

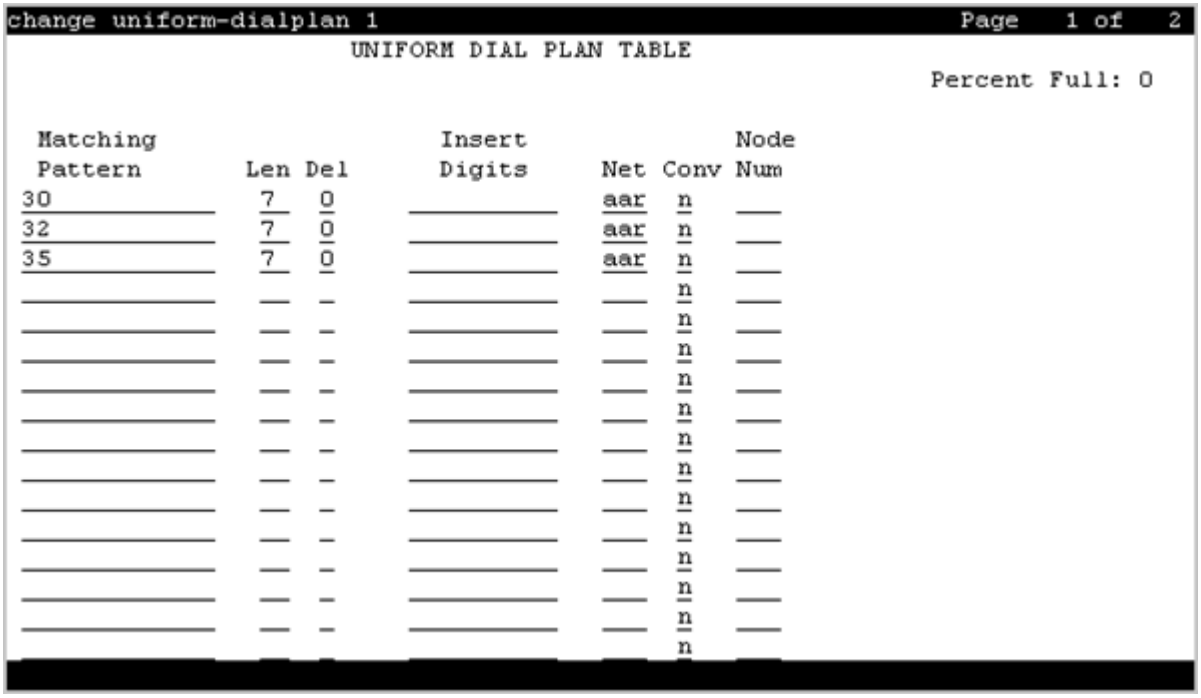
Valid Entry	Usage
96 to 127 blank	The default payload type offered by Communication Manager for SIP trunks. The payload type number encoding for originating (offering) the RFC 2833 RTP “telephone-event” payload format is based on the administered number from this field. This value is used only for Communication Manager originations (outgoing offers). Default is 127.

Chapter 144: Uniform Dial Plan Table

The **UDP** provides a common dial plan length — or a combination of extension lengths — that can be shared among a group of Avaya servers. Additionally, **UDP** can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/ Tributary configurations.

Available only if **Uniform Dialing Plan** is enabled for the system.

Example command: `change uniform-dialplan` *n*, where *n* is the uniform dial plan length.



Related Links

Uniform Dialing Plan on page 806

Conv

Enables or disables additional digit conversion.

Del

Valid Entry	Usage
0 to 9	The number of digits to delete before routing the call. This number must be less than or equal to the number of user dialed digits the system collects to match to the matching pattern.

Related Links

Len on page 940

Insert Digits

Use this field to specify the digits or the number of digits of the location prefix that Communication Manager inserts in the dialed string before routing a call.

Valid entry	Usage
0 to 9	The digits that replace the deleted portion of the dialed number. The deleted portion is the number of digits that Communication Manager deletes specified in the **Del** field. You can enter up to 10 digits. Leave the field blank to delete the digits specified in the **Del** field.
L*x*	The variable *x* is a number from 1 to 11 and represents the number of leading digits of the administered location prefix. Communication Manager prepends the digits of the location prefix to the dialed string. The value of *x* must be less than the value of the **Prefix** field on the Location screen.

Related Links

Locations on page 538

Len

Valid Entry	Usage
1 to 18	The number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the **Matching Pattern** value. The value 2 can be used only when **Insert Digits** contains an Lx value.

Related Links

Insert Digits on page 940
Matching Pattern on page 941

Matching Pattern

Communication Manager matches the value of the **Matching Pattern** field to the dialed numbers.

Valid entries	Usage
0 to 9	Communication Manager matches the value with the dialed numbers. You can enter up to 13 digits.
blank	Leave the field blank if you do not want to specify a pattern.

Net

Valid Entry	Usage
aar ars enp ext	The server or switch network used to analyze the converted number. The converted digit-string is routed either as an extension number or through its converted AAR address, its converted ARS address, or its ENP node number. For enp, the ENP node number must be administered, digit conversion must be disabled, and the **Insert Digits** must not be administered.

Related Links

Conv on page 939
Insert Digits on page 940
Node Num on page 941

Node Num

Valid Entry	Usage
1 to 999	The ENP (Extension Number Portability) Node Number.

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used. For details on the system memory resources, see *Avaya Aura*® *Communication Manager System Capacities Table*, 03-300511.

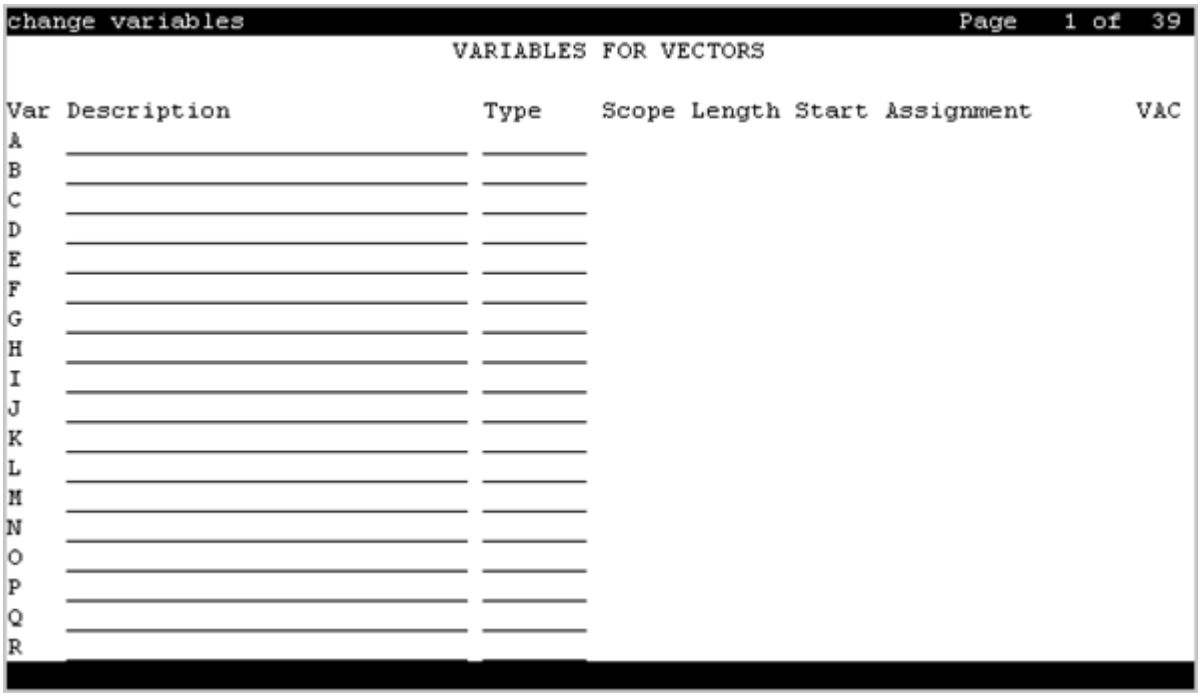
Chapter 145: User Profile

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*, 03-300431. For more information on administering user profiles and logins, see AAA Services in *Avaya Aura® Communication Manager Feature Description and Implementation*, 555-245-205.

Chapter 146: Variables for Vectors

This screen creates variables and defines the necessary parameters for each variable type. Specifies the variable type, the name to use for the variable, the size of the variable, how the variable gets set or assigned, and whether the variable is local or global. Up to 702 variables can be supported using A to Z and AA to ZZ rows.

Example command: `change variables`



Assignment

A number to pre-assign to the variable. This field displays the current value for global values. Administration is optional. Available only when the type is value or collect G .

Related Links

Type on page 944

Description

An identifying name or description of the vector variable. Administration is optional. Default is blank. Accepts up to 27 characters.

Length

Valid Entry	Usage
1 to 16	The maximum number of digits from the data to assign to the variable. This field does not apply to the doy, dow, and tod variables. This entry is required for all types to which it applies.

Scope

Valid Entry	Usage
G	The variable is used globally.
L	The variable is used locally.

Start

Valid Entry	Usage
1 to 96	Specifies the beginning character position of the data digits string to be used for assigning to the variable. The combination of the **Start** position and maximum length of the digits string defines what is to be assigned to the variable. If the number of digits to be used is less than the maximum length specified, only that portion is assigned to the variable.

Type

Valid Entry	Usage
ani asaiuui collect	The vector variable type.

Valid Entry	Usage
dow doy stepcnt tod value vdn vdntime	

VAC

Valid Entry	Usage
1 to 9 blank	The Vector Variable Feature Access Code (FAC) to use for changing the value.

Var

Valid Entry	Usage
A to Z, AA to ZZ	The letter identifying the row of a specific vector variable.

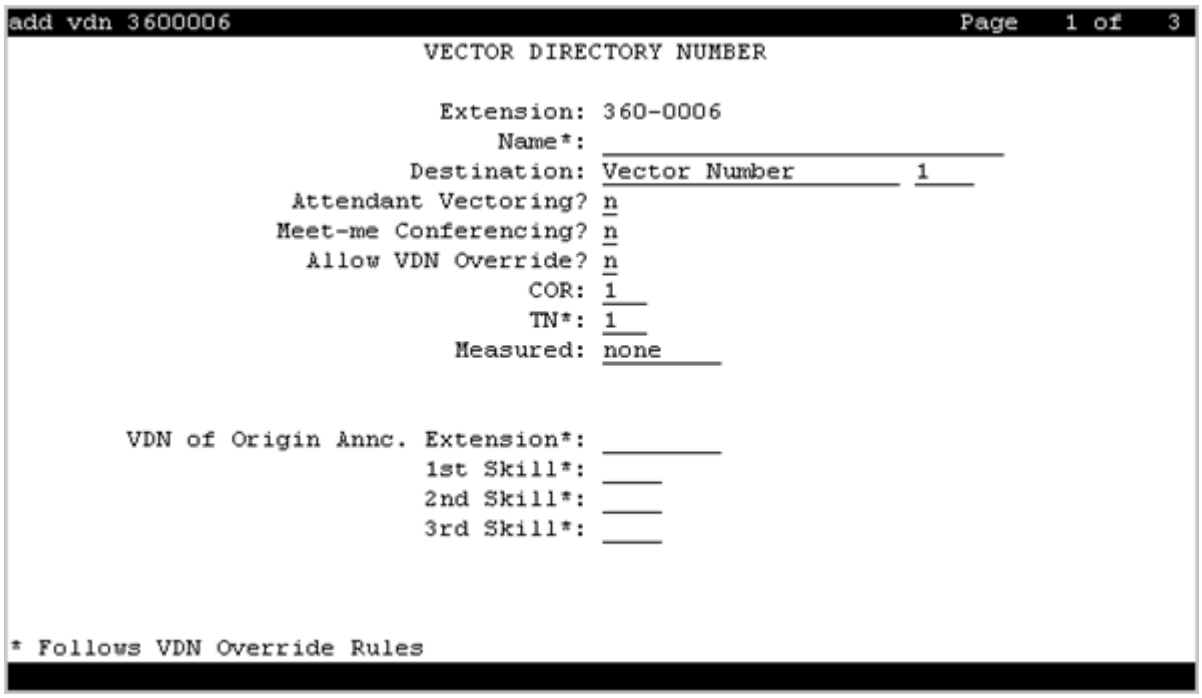
Chapter 147: Vector Directory Number

Defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector. VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed through direct dial local telephone company central office trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN can be the Night Destination for a LDN.

An asterisk (*) next to a field indicates that the field follows VDN override rules when the system changes the active VDN for a call.

Example command: `add vdn` *n*, where *n* is the VDN extension number.

Vector Directory Numbers: page 1



Comments on this document? infodev@avaya.com

1st/2nd/3rd Skill

This field follows VDN override rules when the system changes the active VDN for a call. Available only if **Expert Agent Selection (EAS)** is enabled and **Meet-me Conferencing** is disabled.

Valid entry	Usage
1 to 8000	The skill number for each skill.
blank	Not administered. This is the default.

Related Links

Expert Agent Selection (EAS) on page 808
Meet-me Conference on page 949

Acceptable Service Level (sec)

Available only if **BCMS/VuStats Service Level** is enabled for the system and the VDN is measured by BCMS.

Valid Entry	Usage
0 to 9999	The number of seconds within which calls to this VDN should be answered. This allows BCMS to report the percentage of calls that were answered within the specified time.
blank	Not administered. This is the default.

Related Links

BCMS/VuStats Service Level on page 807
Measured on page 948

Allow VDN Override

Available only if **Meet-me Conferencing** is disabled for that particular VDN

Valid Entry	Usage
y	Allows the system to change the routed-to VDN to the “active” VDN for a call. The “active” VDN is the VDN to be used for parameters associated with the call such as VDN name, skills, tenant number, BSR application, VDN variables, and so on. The field follows VDN override rules when the system changes the “active” VDN for a call.
n	The routed-to VDN does not become the active VDN. The parameters of the original VDN are used. This is the default.

Related Links

Meet-me Conference on page 949

Attendant Vectoring

Indicates whether or not Attendant Vectoring is selected on this VDN. Attendant Vectoring does not support Call Center features.

Related Links

Attendant Vectoring on page 796

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) of the VDN consisting of a one- or two-digit number. This field cannot be blank.

Destination

Specifies if the calls are routed using a Vector Number or Policy Routing Table.

Extension

The extension number of the VDN. The extension is a number that starts with a valid first digit and length as defined by the system's dial plan.

Measured

Enables measurement data collection for this VDN. Data can be collected for reporting by BCMS or the Call Management System. Available only if **Meet-me Conferencing** is disabled. **BCMS** must be enabled for the system for this field to be set to internal or both. In addition, the appropriate CMS release must be administered for the system if this field is external or both.

Valid Entry	Usage
internal	Data is measured internally by BCMS.
external	Data is measured internally by CMS.
both	Data is measured internally by both BCMS and CMS.
none	Data is not measured. This is the default.

Related Links

Meet-me Conference on page 949
BCMS (Basic) on page 807

Meet-me Conference

Enables or disables Meet-me Conferencing for this VDN. Determines if the VDN is a Meet-me Conference VDN.

Note:

If the VDN extension is part of your DID block, external users are able to access the conference VDN. If the VDN extension is not part of the DID block, only internal callers on the network (including DCS or QSIG) or remote access callers can access the conference VDN.

Available only if **Enhanced Conferencing** is enabled for the system. If **Enhanced Conferencing** is enabled, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Related Links

Enhanced Conferencing on page 799

Name

The name associated with the VDN. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. Accepts up to 27 alphanumeric characters.

The name might be truncated on agents’ displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent.

Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphone, and 9600-Series H.323 Deskphone. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information on the 2420J, see *2420 Digital Telephone User's Guide*.

Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, are displayed incorrectly on a BRI station.

Related Links

Display Character Set on page 787
Meet-me Conference on page 949

Number

Valid Entry	Usage
1 to 8000	The number of the vector or the Policy Routing Table (PRT) to which the calls are allocated.

Service Objective

Use this field to assign a service level to the VDN. The system displays this field only if you set the **Dynamic Advocate** field to y. The range of the acceptable value is between 1 to 9999. The default value is 20. For more information on **Service Objective**, see *Avaya Business Advocate User Guide*.

Related Links

BCMS/VuStats Service Level on page 807
Dynamic Advocate on page 808

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

VDN of Origin Annc. Extension

The extension number of the VDN of Origin announcement. A VDN of Origin announcement is a short recording that identifies something about the call originating from the VDN. The agent hears the recording just prior to the delivery of the call. This field follows VDN override rules when the system changes the “active” VDN for a call.

Available only if VDN of **Origin Announcement** is enabled for the system and **Meet-me Conferencing** is disabled.

Related Links

Meet-me Conference on page 949
VDN of Origin Announcement on page 812

Vector Number

An identifying number that specifies the call vector that is accessed through the VDN. This field cannot be blank.

Vector Directory Number: page 2 (Meet-me Conference disabled)

```
add vdn 3600006                                          Page   2 of   3
                          VECTOR DIRECTORY NUMBER

                            AUDIX Name: _______
                     Return Destination*: _______________
                 VDN Timed ACW Interval*: ____ After Xfer or Held Call Drops*? n
                       BSR Application*: ___
           BSR Available Agent Strategy*: 1st-found      Used for BSR Polling? n
                       BSR Tie Strategy*: system

                  Observe on Agent Answer? n

    Send VDN as Called Ringing Name Over QSIG? n

                Display VDN for Route-To DAC*? n
              VDN Override for ASAI Messages*: no

                        BSR Local Treatment*? n

                Reporting for PC or POM Calls? n
              Pass Prefixed CPN to VDN/Vector*? system
* Follows VDN Override Rules
```

Related Links

Meet-me Conference on page 949

After Xfer or Held Call Drops

Use this field to put an auto-in agent into the Timed ACW mode when an ACD or DAC call on hold drops or the call is transferred and the Timed ACW feature is active. This replaces the Available mode.

Valid Entry	Usage
y	Communication Manager places an agent in the Timed ACW state if a caller on hold disconnects or the call is transferred, in addition to the existing conditions for Timed ACW.
n	Communication Manager continues with the existing operation. This is the default value.

AUDIX Name

The name of the AUDIX machine if this VDN is associated with the AUDIX vector.

BSR Application

Available only if **Lookahead Interflow (LAI)** and **Vectoring (Best Service Routing)** are enabled for the system. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Valid Entry	Usage
1 to 511 blank	A one- to three-digit number that specifies an application plan for the VDN when using multi-site Best Service Routing with this VDN.

Related Links

Lookahead Interflow (LAI) on page 809
Vectoring (Best Service Routing) on page 811

BSR Available Agent Strategy

This field determines how Best Service Routing (BSR) identifies the best split or skill in an agent surplus situation to route a call. The system displays this field only if **Vectoring (Best Service Routing)** is enabled. If **Meet-me Conferencing** is disabled, the **BSR Available Agent Strategy** field follows VDN override rules when the system changes the active VDN for a call.

Valid entry	Usage
1st-found	The call is routed to the first available agent.
UCD-LOA	The call is routed to the least occupied agent, without regard to skill level. This option is available only if **Least Occupied Agent (LOA)** or **Business Advocate** is enabled.
UCD-MIA	The call is routed to the most idle agent, without regard to skill level. This option ensures a high degree of equity in agent workloads even when call-handling times vary.
EAD-LOA	The call is routed to the highest skill level agent with the lowest occupancy. This option is available only if **Expert Agent Selection**, and **Least Occupied Agent (LOA)** or **Business Advocate** are enabled.
EAD-MIA	The call is routed to the most idle agent with the highest skill level. This option is available only if **Expert Agent Selection (EAS)** is enabled.

Related Links

Business Advocate on page 808
Expert Agent Selection (EAS) on page 808
Least Occupied Agent on page 809
Vectoring (Best Service Routing) on page 811

BSR Local Treatment

Enables or disables local server control of BSR treatment. In a multi-site BSR configuration, a call that arrives at a local communication server can be rerouted to a remote server located in a different part of the world. This feature maintains control at the local server and allows the server to provide local audio feedback for IP and ISDN calls, or to take back the call while the call waits in queue on a remote server. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Note:

This field must be enabled on both the local and remote vdns, or else call interflow attempts might result in dropped calls.

BSR Tie Strategy

Available only if **Vectoring (Best Service Routing)** is enabled for the system. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Valid Entry	Usage
system	System-wide settings apply.
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Related Links

Vectoring (Best Service Routing) on page 811

Call Origin

The **Call Origin** field is applicable only if you enter a VDN extension in the **Return Destination** field. Use this field to administer the types of calls that will be redirected to the extension in the **Return Destination** field when the destination is released.

Valid Entry	Usage
external	Applies VDN Return Destination only to incoming external trunk calls directly to the VDN. This is the default setting.

Table continues…

Valid Entry	Usage
internal	Applies VDN Return Destination only to internal calls that have not been through vector processing. Internal calls for this purpose include trunk calls that are transferred, forwarded, or covered to the VDN or ASAI routed.
both	Applies VDN Return Destination to both internal and external calls.

Display VDN for Route-To DAC

Enables or disables the display of the VDN when a call arrives as a result of a route-to at an agent terminal under the following conditions:

- A route-to number with coverage is enabled, or route-to digits with coverage is enabled and the vector command routes a call to an agent as an EAS direct agent call.
- Adjunct routing routes a direct agent call to the agent.

When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Available only if **Expert Agent Selection (EAS)** is enabled for the system.

Related Links

Expert Agent Selection (EAS) on page 808

Observe on Agent Answer

Enables or disables a service observer to start observing a call to the VDN when the call is delivered to the agent or station. The ability to observe VDNs when the call is delivered to an agent/station is activated by setting the **Observe on Agent Answer** field to `y`. The observer sees the name of the VDN, agent, or trunk as each is accessed in sequence by the VDN. For example, during vector processing the VDN name is displayed, but when the call connects to an agent, the agent name is displayed. When the observer connects to a call in vector processing, the system maintains the connection until the call is disconnected or the observer disconnects, even if the call is routed or transferred externally. If the observer does not disconnect after one observed call is disconnected, the observer is connected to another call on the same VDN. Observing is listen only as long as the call is in vector processing. Once the call is out of vector processing, an observer with listen/talk capability can talk as well as listen.

Note:

When the **Observe on Agent Answer** field is set to `y`, service observers of the VDN are not connected to a call until the call is delivered to an agent. The default setting of this field is `n`. With the default setting, the non-by location observer of the VDN can hear the call throughout the vector processing and delivery to the agent. For information on **Observe on Agent Answer**, see *Avaya Aura® Call Center Elite Feature Reference*.

Pass Prefixed CPN to VDN/Vector

This feature is used to optionally pass the CPN prefix to VDNs and vectors as part of the CPN. This option can be set on a system wide basis, and overridden, if required, on an individual VDN screen. Accordingly, international call handling can be improved by prioritizing and routing more intelligently, such as by automatically routing calls to agents who speak the pertinent language. For example, if a customer from Germany calls in to a call center in Spain, based on the international code of Germany (0049 - international prefix + country code), the call can be routed to a German speaking agent. Also, an international call might be given priority over a national call to save toll charges.

PC Predictive Reports Skill

The **PC Predictive Reports Skill** field is applicable only if the **ASAI Link Plus Capabilities** or the **Computer Telephony Adjunct Links** field is enabled on page 3 of the System-Parameters Customer-Options screen, and if you enable the **Reporting for PC or POM Calls** field on page 2 of the Vector Directory Number screen.

Valid Entry	Usage
1 to 99	Skill number for S8300 switches. VDN option applies to PC reporting.
1 to 8000	Skill number for all other platforms. VDN option applies to PC reporting.
blank	VDN option applies to Proactive Outreach Manager (POM) calls.

Related Links

Reporting for PC or POM Calls on page 955
ASAI Link Plus Capabilities on page 795
Computer Telephony Adjunct Links on page 797

Reporting for PC or POM Calls

The **Reporting for PC or POM Calls** field indicates that the VDN is used for POM or Proactive Contact (PC) outbound calling. If you have switch-classified outbound calling with PC, use this field and the associated skill field to activate the improved integration with PC feature. If you have POM, use the field to activate this VDN for indication to reporting of a Proactive Outreach Manager call to the VDN. Valid entries are y and n. The default is n.

Note:

You can administer the field only after you set the **Measured** field on page 1 of the Vector Directory Number screen to either external or both.

For more information on how to use the feature, see Proactive Contact outbound calling improved reporting in the *Avaya Aura® Call Center Feature Reference* document.

Related Links

Measured on page 948

Return Destination

The VDN extension number to which an incoming trunk call is routed if it returns to vector processing after the agent drops the call. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Send VDN as Called Ringing Name Over QSIG

Valid Entry	Usage
y	If the ISDN trunk call is made, receiver can view the VDN name when the telephone is ringing.
n	The receiver cannot view the VDN name when the telephone is ringing. The default value is n.

Used for BSR Polling

Use the **Used for BSR Polling** field to prevent Communication Manager from sending messages to CMS or Avaya IQ regarding unmeasured calls for unmeasured VDNs that route to vectors with the **`BSR reply-best`** command.

The **Used for BSR Polling** field is applicable to unmeasured VDNs assigned to a vector that has consider steps terminated by the reply-best step and no-queuing step. The trunk groups that are used to route to the VDNs are also unmeasured.

For **Used for BSR Polling** to function, set the **Measured** field to none.

Valid entry	Usage
y	Communication Manager does not send messages for unmeasured calls to CMS or Avaya IQ.
n	Communication Manager sends messages, even for unmeasured calls, to CMS or Avaya IQ. The default value is n.

VDN Override for ASAI Messages

Determines if the active VDN is sent as the called number for ISDN Trunk ASAI messages. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. The “active” VDN is the VDN receiving the call and will be changed to a

routed-to VDN if **Allow VDN Override** is enabled. Available only if **ASAI Link Core Capabilities** is enabled for the system.

Valid Entry	Usage
n	The “Called Number” information is sent for the “Call Offered”, “Alerting”, “Queued”, and “Connect” ASAI event notification messages. The adjunct-request message is always the called VDN extension in the Called Number IE sent in the incoming ISDN SETUP message or the local call’s called number and does not change after routing to the called VDN and subsequent routed-to VDNs.
ISDN Trunk	When an incoming ISDN trunk call is routed to this VDN, the “Called Number” information sent in the ASAI event and “Adjunct Route Request” ASAI messages is the “active VDN” extension. This extension becomes associated with the call based on the VDN Override rules. This option does not apply to local/internal calls.
all	The active VDN is used for the called number for all types of calls to the VDN, including local/internal calls as well as external incoming ISDN trunk calls.

Related Links

Allow VDN Override on page 947
ASAI Link Core Capabilities on page 795

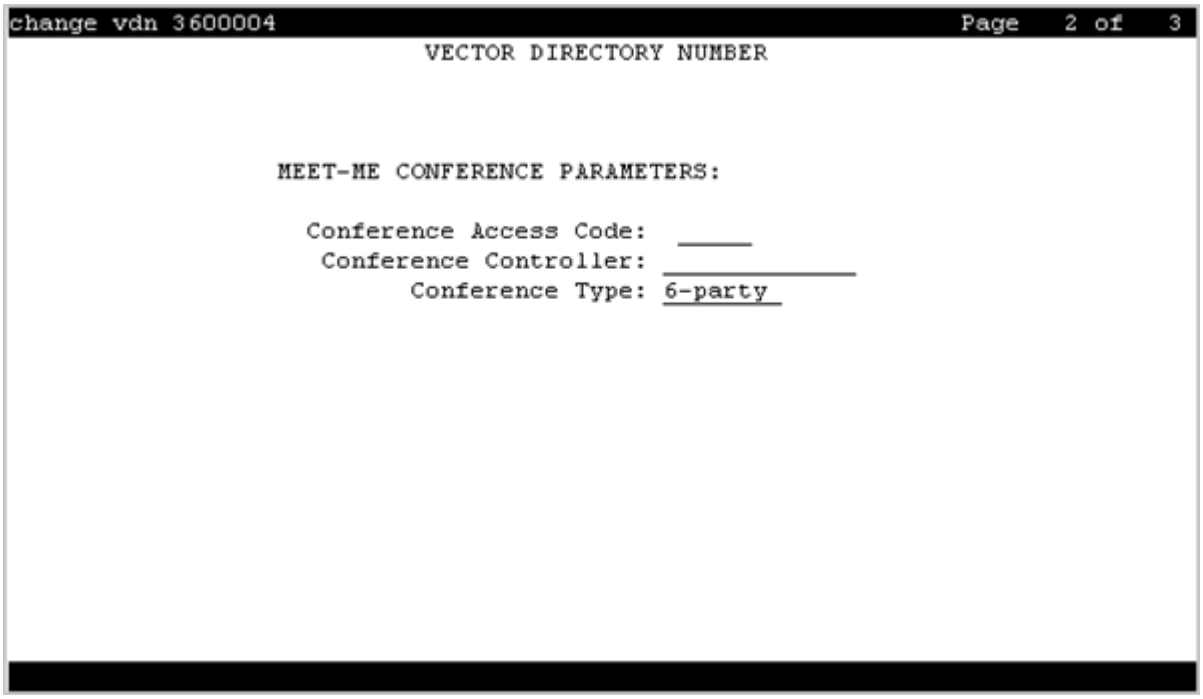
VDN Timed ACW Interval

Valid Entry	Usage
1 to 9999 blank	Sets the length of the timed ACW Interval. When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field takes precedence over the hunt group **Timed ACW Interval**. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Related Links

Timed ACW Interval (sec) on page 384

Vector Directory Number: page 2 (Meet-me Conference enabled)



Related Links

Meet-me Conference on page 949

Conference Access Code

Security alert:

To ensure conference security, always assign an access code to a Meet-me Conference VDN.

The Meet-Me Conference VDN access code. Accepts up to six digits.

Conference Controller

Controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in from remote access trunks.

Valid Entry	Usage
Extension number	Only a user at this extension can change the access code for that VDN using a feature access code.

Table continues…

Comments on this document? infodev@avaya.com

Valid Entry	Usage
blank	Any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Conference Type

The conference type that is appropriate for the call.

Valid Entry	Usage
6-party	For six or fewer participants. Default is 6-party.
expanded	Enables the Expanded Meet-me Conference feature. For a conference with more than six participants.

Route-to Number

The ARS or AAR Feature Access Code (FAC) followed by the routing digits. Alternately, the unique UDP extension. Accepts up to 16 digits. Allows administration of the routing digits (the ARS/AAR Feature Access Code with the routing digits and the Conference ID digits for the VDN). Available only for expanded conference types.

Note:

The Route-to Number must be unique across all Expanded Meet-me Conference VDNs.

Related Links

Conference Type on page 959

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Vector Directory Number: page 3

```
add vdn 3600006                                          Page   3 of   3
                          VECTOR DIRECTORY NUMBER

                                VDN VARIABLES*

                     Var  Description       Assignment
                     V1   _______________   _______________
                     V2   _______________   _______________
                     V3   _______________   _______________
                     V4   _______________   _______________
                     V5   _______________   _______________
                     V6   _______________   _______________
                     V7   _______________   _______________
                     V8   _______________   _______________
                     V9   _______________   _______________

                     VDN Time-Zone Offset*: + 00:00
                     Daylight Saving Rule*: system
 Use VDN Time Zone For Holiday Vectoring*? n
    Apply Ringback for Auto Answer calls*? y

* Follows VDN Override Rules
```

VDN VARIABLES

When **Meet-me Conferencing** is disabled, the system displays an asterisk (*) next to the heading, indicating that variables V1 through V9 follow VDN override rules when the system changes the “active” VDN for a call.

Related Links

Meet-me Conference on page 949

Assignment

Valid Entry	Usage
0 to 9	Assigns an up to 16-digit unvalidated decimal number to each of the VDN variables V1 through V5.
blank	No decimal number is assigned to the VDN variable.

Description

A description of the VDN variable. Accepts up to 15 characters.

Var

The number assigned to the VDN variable.

Apply Ringback for Auto Answer calls

The **Apply Ringback for Auto Answer calls** field is applicable only if you enable the **Auto Answer** field on either page 2 of the Station screen or on page 1 of the Agent Login ID screen. You can use the field to not apply ringback to the callers for calls delivered to agents in the auto answer mode when application of ringback is undesirable. This field follows VDN Override rules and the setting on the active VDN for the call is applicable.

Valid Entry	Usage
y	Applies audible ringback to the callers during any delay experienced for calls delivered to agents in the auto answer mode. This is the default value.
n	Prevents returning audible ringback to the callers for calls delivered to agents in the auto answer mode.

Related Links

Auto Answer on page 44
Auto Answer on page 704

Daylight Saving Rule

Defines the daylight saving time rule. The daylight saving time rule is applied to `goto time-of-day` commands in the vector that is assigned to the VDN. The time-of-day calculations are based on the local time of the receiving call's VDN. The assigned rule number applies start and stop rules that are administered for the system for that rule.

Valid Entry	Usage
system	The system uses the same daylight saving time rule as the system clock.
0	No daylight saving rule is applied. If the system time has a daylight saving rule specified, this rule is removed before evaluating the `goto if time-of-day` conditional.
1 to 15	The **Daylight Saving Rule** number. When you use a number other than 0, the rule associated with the main server clock display time and the main server offset are not used. The offset and rule assigned to the active VDN for the call are applied to the operating system standard time so that local time for the VDN is used to test the time-of-day step.

Related Links

Daylight Saving Rules on page 192
VDN Time Zone Offset on page 962

Use VDN Time Zone For Holiday Vectoring

Valid Entry	Usage
y	Communication Manager uses the VDN time zone offset and DST adjustment for processing the date and time assigned in the Holiday Table screen.
n	Communication Manager uses the system time for processing the date and time assigned in the Holiday Table screen. This is the default value.

VDN Time Zone Offset

Valid Entry	Usage
+, - 00-23 - hour 00-59 - minute	This field is applied against the switch clock when a time of day vector command is executed. Daylight saving time changes are handled by the switch clock using the existing operation. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. The valid entries are based on a syntax of +HH:MM. The default is +00:00. When the default is set, the system switch time is used without modification.

Related Links

Meet-me Conference on page 949

Chapter 148: Video Bridge

Used to configure available ad-hoc conferencing resources. For more detailed information on Avaya Video Telephony, see *Avaya Video Telephony Solution Networking Guide, 16-601423.*

Example command: `add video-bridge n`, where *n* is the video bridge number.

Bridge ID

The ID number for this video bridge.

Call Rate

The maximum allowable call rate for the conference. Available only when **Far End Resource Info** is disabled and **Type** is not administered.

Related Links

Far End Resource Info on page 963
Type on page 965

Far End Resource Info

Enables or disables far end resource tracking and reporting.

Valid Entry	Usage
y	The far end tracks port usage and provides updates on resource availability.
n	No resource information is provided from the far end.

ID Range Start/End

A range of conference IDs that this video bridge can use. There must be enough IDs so that all ports can be used — one ID for every six ports. Accepts up to nine digits.

The default is blank.

Available only for h.323 trunk groups.

Related Links

Group Type on page 482

Max Ports

Valid Entry	Usage
3 to system max	The maximum number of video conferencing ports for this video bridge. Default is none.

Name

The name that identifies this video bridge. Accepts up to 30 alphanumeric characters.

Priority Factory Number

Priority versus Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates the conference, the **Priority Factory Number** is used. **Priority Factory Number** can have a dedicated video bridge or a bridge with bandwidth more than the **Standard Factory Number**. Available only for h.323 or sip trunk groups when the **Far End Resource Info** is enabled.

Valid Entry	Usage
1 to 9 digits (0,9)	At least one of **Priority Factory Number** or **Standard Factory Number** must be filled in. Standard and Priority factory numbers can be the same.
blank	Priority calls can use the bridge, but prefer a bridge with a priority factory. This is the default.

Related Links

Far End Resource Info on page 963
Group Type on page 482

Standard Factory Number on page 965

Standard Factory Number

Priority versus Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates the conference, the **Priority Factory Number** is used. **Priority Factory Number** has a dedicated video bridge or a bridge with better bandwidth than the**Standard Factory Number**. Available only for h.323 or sip trunk groups. For h.323, the **Far End Resource Info** must be enabled.

Valid Entry	Usage
1 to 9 digits (0,9)	At least one of **Priority Factory Number** or **Standard Factory Number** must be filled in. Standard and Priority factory numbers can be the same.
blank	Non-priority conferences are unable to use this video bridge. Default is blank.

Related Links

Priority Factory Number on page 964
Far End Resource Info on page 963
Group Type on page 482

Trunk Groups

Valid Entry	Usage
1 to 2000 blank	Assigns trunk groups to this video bridge. At least one incoming and one outgoing trunk, or a two-way trunk, must be assigned. All trunks on a given video bridge must be the same type; H.323 and SIP trunks cannot be mixed. The default is blank.

Related Links

Group Type on page 482

Type

The type of video bridge. Can be administered for existing or new video bridges. Available only if **Far End Resource Info** is disabled.

Valid Entry	Usage
Exact	All participants in the conference must use the exact rate specified or be in the audio-only mode.
Maximum	**Call Rate** is enabled, and any call rate up to the configured rate is allowed in a conference. Default is Maximum.
Region	The initial entry for an existing video bridge. **Call Rate** is unavailable.

Related Links

Call Rate on page 963
Far End Resource Info on page 963

Chapter 149: Virtual MAC Addresses

Lists the virtual Media Access Control (MAC) addresses on the system.

Example command: `display virtual-mac-address n`, where *n* is the virtual MAC addresses table number.

MAC Address

Virtual MAC address shared by duplicated TN2602AP circuit packs. Accepts up to 15 alphanumeric characters.

Note:

The 4606, 4612, and 4624 telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from archive to standby media processor is in process, calls might be dropped.

Used

Indicates whether or not the associated virtual MAC address has been assigned in the system.

Appendix A: PCN and PSN notifications

PCN and PSN notifications

Avaya issues a product-change notice (PCN) if any software update. For example, a PCN must accompany a service pack or a update that must be applied universally.Avaya issues product-support notice (PSN) when there is no update, service pack, or release fix, but the business unit or services need to alert Avaya Direct, BusinessPartners, and customers of a problem or a change in a product. A PSN can also be used to provide a work around for a known problem, steps to recover logs, or steps to recover software. Both these notices alert you to important issues that directly impact Avaya products.

Viewing PCNs and PSNs

To view PCNs and PSNs, perform the following steps:

Procedure

1. Go to the Avaya Support website at http://support.avaya.com.

   Note:

   If the Avaya Support website displays the login page, enter your SSO login credentials.

2. On the top of the page, click **DOCUMENTS**.
3. On the Documents page, in the **Enter Your Product Here** field, enter the name of the product.
4. In the **Choose Release** field, select the specific release from the drop-down list.
5. Select the appropriate filters as per your search requirement. For example, if you select Product Support Notices, the system displays only PSNs in the documents list.

   Note:

   You can apply multiple filters to search for the required documents.

Signing up for PCNs and PSNs

Manually viewing PCNs and PSNs is helpful, but you can also sign up for receiving notifications of new PCNs and PSNs. Signing up for notifications alerts you to specific issues you must be aware of. These notifications also alert you when new product documentation, new product patches, or new services packs are available. The Avaya Notifications process manages this proactive notification system.

To sign up for notifications:

Procedure

1. Go to the Avaya Support Web Tips and Troubleshooting: E-Notifications Management page at https://support.avaya.com/ext/index?page=content&id=PRCS100274#.
2. Set up e-notifications.

   For detailed information, see the **How to set up your E-Notifications** procedure.

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